TAPR HF Special Interest Group Archive, January 1996 through December 1996

January 1996 through December 1996

TAPR's forum for experimentation and development of HF digital communications, including modulation, coding, protocols, DSP platforms, modems, propagation, and on-air testing.

This volume contains 11 original monthly files and 952 messages, arranged chronologically. The PDF is a readable access copy: MIME transfer encodings are decoded, HTML bodies are converted to plain text, and non-text attachments are represented by descriptive notes. The raw source text is supplied separately with the Internet Archive item.

Month	Messages	Original source file
1996-01	57	hfsig.9601
1996-02	83	hfsig.9602
1996-03	75	hfsig.9603
1996-04	31	hfsig.9604
1996-06	84	hfsig.9606
1996-07	149	hfsig.9607
1996-08	124	hfsig.9608
1996-09	28	hfsig.9609
1996-10	136	hfsig.9610
1996-11	61	hfsig.9611
1996-12	124	hfsig.9612

January 1996

Original source file: hfsig.9601
Source: https://archive.org/details/tapr-1997
Recovered from: https://archive.org/download/TAPR-HF_1996-01/TAPR-HF_1996-01.txt

1. Unsubscription help?

From: Sivula Timo <sivula@ncsnrs01es.ntc.nokia.com>
Date: Tue, 02 Jan 96 07:18:00 eet
To: "'HFSIG'" <hfsig@tapr.org>
Message-ID: <30E8C150@ntcite01es.ntc.nokia.com>
Rendered body type: text/plain

Hello,

I have tried to unsubscribe from this group for a coupl eof weeks now without any
luck. This datapoint.com thing doesn't seem to exist when I send the unsubscribe
messahes for the listserv.

The problem is that our locla server has chnaged, and the messages form hfsig go
to this old server, and I am not receiving them. My idea was to unsubscribe from the
list and then re-subscribe with the new adress in order to be able to follow the interesting
discussions going on.

Could somebody give me a hand and mail me how to do this. Please, do not reply to the
hfsig group as I cannot receive any mails from there before I have unsubscribed ....

73.

Timo.Sivula@ntc.nokia.com

2. Re: Nordic HF Conference

From: "Bergzen Hakan, KARL" <Hakan.Bergzen@telub.se>
Date: Tue, 02 Jan 96 14:45:00 GMT
To: HFSIG at TAPR <hfsig@tapr.org>
Message-ID: <30F44269@noak>
Rendered body type: text/plain

Hello Rich, Charles and all others,
I saw my name coming up in this newsgroup, forcing me out of my safe Nordic seclusion :-)
The Nordic HF homepage deals with the Nordic HF conferences held every third year since 1986. The homepage was established in Nov 95. The address is:
  http://www.telub.se/Radio/NordicHF/
I plan to make a small revision in near-time: correcting some errors, including a page on HF modems (.au files), and a page on the ALE standard and the HFIA group.
When I included the zipped postscript files I tested printing every one of them. I succeded in all but one (a paper by G Andersson, i.e. not the one Rich refer to). Colleagues have also tried and succeded. The only error cause I can think of is, since you succeded in downloading and extracting the postscript files, that your postscript printer generates a printing timeout error. The paper by Patrik Nilsson you refer to has a large Windows snapshot picture on page six which may cause this. Try hooking up your printer directly to the parallell port of your PC, thus omitting LAN delays.
And Rich, if this does not solve your problem please send me your snail-mail address and I will send you Patrik's paper directly.

Please E-mail me (hakan.bergzen@telub.se) if any of you interested have problems accessing the page or have ideas on suitable topics or other comments, and inform
anybody of this homepage that might be interested.

Happy surfing,
Hakan Bergzen (hakan.bergzen@telub.se)

PS The ongoing technical discussions regarding the various modem developments sure is fascinating. Some of you might get some good implementation ideas by studying the available HF modem standards: e.g. MIL-STD-188-110A, MIL-STD-188-141A and STANAG 4285.

3. Re: [HFSIG:784] Radio Mail using PACKTOR/GTOR

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 2 Jan 1996 23:18:04 -0800 (PST)
To: FLundgren@daniel.maf.org (Fred Lundgren)
Cc: hfsig@tapr.org
Message-ID: <199601030718.XAA09502@servo.qualcomm.com>
Rendered body type: text/plain

Fred,

Ordinarily I'd say "just figure out a standard for encapsulating IP
in your physical layer (TOR)" and run TCP/IP over the whole thing".

That's still in general a good approach; the ease with which this can be
done with most link media has certainly made the Internet a success.

But HF radio is so slow by comparison to other media that this is not
necessarily a no-brainer; one might consider sending just Internet format
(RFC822) email messages by themselves in a mail-level store-and-forward
network. Similar things have been done in the ham HF network for years.

Phil

---

4. Re: [HFSIG:793] Re: Radio Mail using PACKTOR/GTOR

From: jmorriso@bogomips.com (John Paul Morrison)
Date: Wed, 3 Jan 1996 00:58:22 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <m0tXP1b-000TqUC@bogomips.com>
Rendered body type: text/plain

>
> Fred,
>
> Ordinarily I'd say "just figure out a standard for encapsulating IP
> in your physical layer (TOR)" and run TCP/IP over the whole thing".
>
> That's still in general a good approach; the ease with which this can be
> done with most link media has certainly made the Internet a success.
>
> But HF radio is so slow by comparison to other media that this is not
> necessarily a no-brainer; one might consider sending just Internet format
> (RFC822) email messages by themselves in a mail-level store-and-forward
> network. Similar things have been done in the ham HF network for years.

UUCP is still a good (as far as non TCP/IP methods go) way of moving
mail and news. I recall hearing cc:Mail mentioned, and cc:Mail does
have a UUCP product.

I don't know how AMTOR/PACTOR works; but if you can find a way to run
an 8 bit serial connection over it, UUCP-g protocol should work. Maybe
"UUCP Spoofing" could be done in the HF modem, to make it a bit more
efficient, or easier to plug in to off the shelf products like
cc:Mail, UUPC (actually ccMail's UUCP is just a modified version of
UUPC), etc. If you can do without a commercial product, Taylor UUCP
(which has source code freely available) could be hacked to work with
whatever the HF modem wants to see.

```
------------------------------------------------------------------------
BogoMIPS Research Labs  --  bogosity research & simulation  --  VE7JPM  --
  jmorriso@bogomips.com  ve7jpm@ve7jpm.ampr.org  jmorriso@ve7ubc.ampr.org
------------------------------------------------------------------------
```

---

5. Spread spectrum question

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Thu, 4 Jan 1996 09:52:06 -0800
To: HFSIG@tapr.org
Message-ID: <630D39D2D60@frl.orst.edu>
Rendered body type: text/plain

Hi all,

Here is a practical question to you SS experts:

I have been experimenting with Pawel's latest slow speed BPSK modem. It
looks like a nice bit of work - has many fine attributes that I like
to see in a HF modem, except its weakness due to being very narrow band. If
it was possible to use a SS technique to utilize for example a voice band
bandwidth, it may even be a nice way to learn a couple of new tricks, thus
my questions. Note that everything is inteneded to function at baseband:

1) Could someone explain how to compute the chip rate to spread a digital signal with a symbol rate of say, X, to occupy a bandwidth, Y. Lets assume we want to implement fast frequency hopping (FFH).

2) I would appreciate some explanation and pointers to a suitale P-N sequence generator, like for example Gold sequences (This is to ensure that the resultant spectrum is nice clean, symmetric bell-shaped).

3) I can see how a pair of operators can achieve the initial synchronisation, but I'm looking for ideas on how to keep the chip rate synchronised. For example, if it is possible to achieve and hold initial synchronization - would it be possible/practical to phase lock on to the data rate using a robust (simple) method such as for example early-late gating? What kind of clock stability would be needed?
What alternatives do we have?

Thanks in advance,

--Johan

6. Re: [HFSIG:795] Spread spectrum question

From: danie.brynard@pixie.co.za (Danie Brynard)
Date: Thu, 4 Jan 1996 21:35:34 +0200
To: hfsig@tapr.org
Message-ID: <199601041935.VAA05852@f15.pix.za>
Rendered body type: text/plain

Hi Johan. There are certainly lots of SS buffs around but I can tell you only this:

processing gain = 10*log(spreading width/information bandwidth)

ie for 10kbps at a chipping rate of 1MHz the processing gain is :

10*log(1e6/10e3) = 20dB

So now you can work out from this what the chipping rate must be for the slow BPSK stuff:

say datarate is 10Hz and voice bandwidth is 3kHz. Thus the processing gain is 10*log(3e3/10)=24dB ! Very good I think. But a chipping rate of 3kHz would produce a spectrum 6kHz wide from first null to first null or not ?

So lets take 1.5kHz then, thus processing gain is :12dB Would this be any helpfull ?

Is this correct ? Lets see what the experts say. I like your line of thinking Johan :-)

73 danie zs6awk

7. Re: [HFSIG:795] Spread spectrum question

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 4 Jan 1996 23:27:00 -0800 (PST)
To: FORRERJ@frl.orst.edu
Cc: hfsig@tapr.org
Message-ID: <199601050727.XAA29527@servo.qualcomm.com>
Rendered body type: text/plain

>1) Could someone explain how to compute the chip rate to spread a digital
>signal with a symbol rate of say, X, to occupy a bandwidth, Y. Lets assume
>we want to implement fast frequency hopping (FFH).

Well, for direct sequence, you just chip at 1/Y. The symbol rate doesn't matter (as long as it's lower than the chip rate).

With frequency hopping it's a little different. You have to pick some multiple of your channel bandwidth. That sets your spreading bandwidth; the rate at which you hop doesn't affect this over the long term.

>2) I would appreciate some explanation and pointers to a suitale P-N
>sequence generator, like for example Gold sequences (This is to ensure
>that the resultant spectrum is nice clean, symmetric bell-shaped).

"Shift Register Sequences" by Golomb is the standard reference. Any maximal-length linear feedback shift register will do for starters. It's not necessary to worry about Gold codes until you get serious about CDMA. The main advantage of Gold codes is that you can create a

whole family of them with fairly good cross-correlation properties.
(For example, each GPS satellite uses an assigned Gold code so that
receivers can select them individually without regard to the phase
difference between satellites).

>3) I can see how a pair of operators can achieve the initial
>synchronisation, but I'm looking for ideas on how to keep the chip rate
>synchronised. For example, if it is possible to achieve and hold initial
>synchronization - would it be possible/practical to phase lock on to the
>data rate using a robust (simple) method such as for example early-late
>gating? What kind of clock stability would be needed?
>What alternatives do we have?

Actually, acquisition is the hard part, tracking is the easy part.
Early-late gating is the standard approach. To avoid losing performance,
make sure you actually demodulate your data from an on-time reference,
not from the early or late channels.

Phil

8. DSP based sound cards

From: chbrain@dircon.co.uk (Charles Brain)
Date: Fri, 05 Jan 1996 12:29:11 +0000
To: hfsig@tapr.org
Message-ID: <199601051251.AA28381@felix.dircon.co.uk>
Rendered body type: text/plain

Hello All,

Has anyone got any suggestions of a reasonable quality DSP based
Sound Card. I need to reproduce some .WAV files very accurately. I am
trying to re-produce the test tones supplied by NTIA for their ALE test
S/W, a normal cheap SB card doesn't appear to be good enough.

I thought the Orchid card would be suitable but there may be better ones
around.

Seasons greetings

Charles G4GUO.

9. AutoResponse

From: NULL@synapsis.it
Date: Fri, 5 Jan 96 14:18:38 +0100 (CET)
To: hfsig@tapr.org
Message-ID: <199601051418.IAA25095@sys1.tapr.org>
Rendered body type: text/plain

User not reachable. SMTP account disabled

10. AutoResponse

From: NULL@synapsis.it
Date: Fri, 5 Jan 96 15:59:51 +0100 (CET)
To: hfsig@tapr.org
Message-ID: <199601051559.JAA28138@sys1.tapr.org>
Rendered body type: text/plain

User not reachable. SMTP account disabled

11. Re: [HFSIG:798] DSP based sound cards

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Fri, 5 Jan 1996 10:19:28 -0800
To: hfsig@tapr.org
Message-ID: <649487D29F6@frl.orst.edu>
Rendered body type: text/plain

Hello Charles,

Most of the present sound cards that I know of are 16-bit. The AD-
1848 CODEC is used on several as a low-cost solution. However, if you are
looking for better noise figures, look for a card thatuses a quality drop-
in replacement, such as those made by Crystal semi. The quality of
the eventual D-A is very much dependant on how well the analog board
layout, especially decoupling and grounding have been done. This often is
the difference between a high-quality design and a cheap one.

Also look at all the additional gadgets that they add in the audio path
like for example tone controls and audio amplifiers - if they don't use the

best low-noise parts, these often contribute to poor noise performance let
alone introduce distortion products.

There have been several tests of different sound cards by PC Magazine for
example. There you will see the actual tested frequency response curves.
Unfortunately, few of the big selling cards were found to be rated all that
high. There were two, however, if I remember correctly, that had really
excellent audio responses; The Turtle Beach family cards and the IBM
Windsurfer board. I'm sure that these have drivers that will play .WAV
files.

I would suggest looking at those PC Magazine evaluations again to get a
better idea for a high quality off-the-shelf sound card.

Hope this helps.

--Johan

> Received: from SpoolDir by FRL (Mercury 1.21); 5 Jan 96 06:29:51 PST8PDT
> Return-path: <hfsig@tapr.org>
> Received: from sys1.tapr.org by frl.orst.edu (Mercury 1.21);
>   5 Jan 96 06:29:42 PST8PDT
> Received: from sys1.tapr.org (localhost [127.0.0.1]) by sys1.tapr.org (8.7.2/8.7.2) with SMTP id
HAA21973; Fri, 5 Jan 1996 07:04:22 -0600 (CST)
> Date: Fri, 5 Jan 1996 07:04:22 -0600 (CST)
> Message-Id: <199601051251.AA28381@felix.dircon.co.uk>
> Reply-To: hfsig@tapr.org
> Originator: hfsig@tapr.org
> Sender: hfsig@tapr.org
> Precedence: bulk
> From: chbrain@dircon.co.uk (Charles Brain)
> To: hfsig@tapr.org
> Subject: [HFSIG:798] DSP based sound cards
> X-Listprocessor-Version: 6.0 -- ListProcessor by Anastasios Kotsikonas
> X-Comment: Tucson Amateur Packet Radio HF Special Interest Group
> Content-Type: text/plain; charset="us-ascii"
> X-Mailer: Windows Eudora Version 1.4.4
> X-PMFLAGS: 33554560
>
> Hello All,
>
> Has anyone got any suggestions of a reasonable quality DSP based
> Sound Card. I need to reproduce some .WAV files very accurately. I am
> trying to re-produce the test tones supplied by NTIA for their ALE test
> S/W, a normal cheap SB card doesn't appear to be good enough.
>
> I thought the Orchid card would be suitable but there may be better ones
> around.
>
> Seasons greetings
>
> Charles G4GUO.
>
>

12. No Subject

From: jon <jon@g4tsn.demon.co.uk>
Date: Fri, 05 Jan 96 21:28:20
To: hfsig@tapr.org
Message-ID: <8@g4tsn.demon.co.uk>
Rendered body type: text/plain

--

| jon   EMail jon@
| Mail sent via Demon Internet - Full IP for 10/Month Tel:0181 371 1234 |
---
sign off

13. NORDIC HF95 Proceedings

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Tue, 09 Jan 1996 09:42:09 -0800
To: HFSIG@tapr.org
Message-ID: <9601091736.AA22101@ucs.orst.edu>
Rendered body type: text/plain

Hi,

I wonder if anyone have been successful in printing some of the Nordic HF95

proceedings?

I am quite interested in reading the paper on the all-digital HF radio as it contains some really fundamental concepts - kind of pave the way towards doing it the right way. Unfortunately my Postscript printer does not handle A4 paper and my version of a Postscript viewer (Ghostscript 3.51) only views the first two pages.

Thanks a lot.

--Johan, KC7WW

P.S. I am a free man again - thanks for your patience. There are a couple of things that we should pick up the thread again and get the ball rolling. More on that later.

14. Re: [HFSIG:803] NORDIC HF95 Proceedings

From: erik@ve7mdl.ampr.org
Date: Tue, 09 Jan 96 19:31:33 PST
To: hfsig@tapr.org
Message-ID: <21267@ve7mdl.ampr.org>
Rendered body type: text/plain

Johan,

Have you tried to edit the postscript file. You can often change the 'a4' to 'letter' and - if the pages are not too crowded - you can get it talked into sending it to your printer.

Another trick, I have used, is to set the printer up to believe it is printing A4, but using 8.5x11" paper. it can be done on some laser printers.

Of course, life would be so much simpler if we North American followed international standards :-)

73 de VE7MDL   ....Erik.

15. Re: [HFSIG:801] Re: DSP based sound cards

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 10 Jan 1996 01:03:47 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199601100903.BAA20159@servo.qualcomm.com>
Rendered body type: text/plain

A further comment on sound cards.

It's certainly true that few (if any) of the common 16-bit cards come anywhere near the CD-quality audio that they imply. But for the purpose of hooking up to a HF transceiver, even the worst card is plenty good enough. No reason to spend lots of money here.

Phil

PS to the list manager: could somebody *please* fix the remailer to delete the erroneous "Reply-To: hfsig@tapr.org" header it is currently adding??

16. Re: [HFSIG:805] Re: DSP based sound cards

From: chbrain@dircon.co.uk (Charles Brain)
Date: Wed, 10 Jan 1996 19:00:34 +0000
To: hfsig@tapr.org
Message-ID: <199601101923.AA06036@felix.dircon.co.uk>
Rendered body type: text/plain

>A further comment on sound cards.
>
>It's certainly true that few (if any) of the common 16-bit cards come
>anywhere near the CD-quality audio that they imply. But for the
>purpose of hooking up to a HF transceiver, even the worst card is plenty
>good enough. No reason to spend lots of money here.
>
>Phil
Yes you are right,

What is more important is how accurate the Xtal Oscillator is especially on synchronous bit streams aka the message Johan posted a few months ago.

I tried to be clever and got my 8ary modem to constantly hunt for bit sync
rather than getting sync and locking to it. What happen? Well as soon as the
signal faded it re-synced on the noise and started adding extra bits to the
bit stream, this really screwed up the ECC !

Regards Charles.

17. Channel Simulator

From: chbrain@dircon.co.uk (Charles Brain)
Date: Wed, 10 Jan 1996 20:03:48 +0000
To: hfsig@tapr.org
Message-ID: <199601102026.AA09555@felix.dircon.co.uk>
Rendered body type: text/plain

I thought you might find this interesting

snip-------------------------

A Software Implementation of Channel Degradation
Chris Riddle
NTIA ITS.N1

Overview:

The Institute for Telecommunication Sciences (ITS) has produced a
Compact Disk (CD) for
interoperability testing of ALE HF radio systems that comply with Federal
standards 1045A and 1046/1, .
The 46 minute CD, ALE CD-01A, includes all of the required functions and
protocols in FED-STD-1045A,
and several functions in FED-STD-1046/1. The tones on this first CD are
"clean", i.e., they are pure digital
tones with no channel degradation. CD-01A is available as NIST Special
Database 17, as described in the
"readme.1st" file accompanying this report. A set of twelve CDs, which ITS
produced in FY 95, were tones
degraded digitally using the narrowband Watterson model. The channel
degradations include Gaussian
noise, fading and multipath. This paper explores the various methods and
algorithms used for the
implementation of the Watterson model in the digital domain.

Introduction:

Both the clean and the degraded tones are first generated on a
computer, then transferred to a
CD. The program to generate the degraded tones is named ALECALL and was
written in C. When using
ALECALL, it is first necessary to define the ALE tones and specific channel
conditions by writing an ASCII
protocol file. The protocol file format was designed to be very flexible in
order to accommodate a wide
range of ALE calls and channel conditions. The protocol file is the input to
ALECALL. The output of
ALECALL is either a wave or a pcm sound file using two channels, 16 bits,
and a sampling rate of 44.1
SNR.
There are three basic channel degradation modules which are necessary
to generate degraded
channel conditions implementing the Watterson model. The three modules are
Gaussian noise, multipath,
and fading bandwidth. By adding some combination or all three of the modules
and adjusting parameters
for each of the modules, a wide variety of channel conditions can be
realized including CCIR Good, CCIR
Poor, and Gaussian.
There are some inherent limitations with putting ALE tones under
digitally simulated channel
conditions on CD. The sampling rate is the primary limitation of a CD. The
sampling rate of a CD is 44.1
kHz by definition which limits the greatest frequency that can be
represented to the Nyquist frequency of
22.05 kHz. However, for the narrowband channel simulators, this is more
spectrum than necessary.
The software is in a modular form so that future additions or
modifications can be easily made.
Future channel condition models may include new modules such as impulse
noise. Furthermore, the

modules can be updated or expanded independently. The digital modules have more precision and a
wider range in the parameter settings than is currently available in channel simulation hardware. For
example, with ALECALL, it is possible to have a precisely controlled bandwidth from 0 to 9.9 kHz for
Gaussian noise instead of being locked into a single bandwidth.

Gaussian Noise:

There are two variables that define Gaussian noise: bandwidth and signal to noise (SNR) ratio.
ALECALL allows any bandwidth of Gaussian noise ranging from 0 to 9.9 kHz, and any SNR ratio from -20
dB to 80 dB. Gaussian noise is very convenient to utilize because when it is added to any linear system, it
remains Gaussian noise. Therefore, both the bandwidth and the SNR ratio of the Gaussian noise can be
adjusted independently.
There is no standard C call for generating Gaussian noise. However, it is possible to define a
function that closely approximates Gaussian noise. The approximation used in ALECALL for generating
Gaussian noise is as follows:

```
double x = 0;
int k;
for(k=0; k<12; k++)
 x += (double)rand()/RAND_MAX;
x -= 6.0;
```

This produces nearly perfect Gaussian noise with a variance of 1 and a mean of 0. The maximum and
minimum values of this function is 6 and -6 respectively. Ideally, Gaussian noise should be able to take
values greater than 6 and less than -6. However, such values would be greater than 6 standard deviations
from the mean which occurs less than 0.001% of the time.
Once the Gaussian noise is generated, it is then shaped to have the predefined SNR ratio and
bandwidth. The bandwidth is controlled by passing the Gaussian noise through a fourth order Butterworth
low pass infinite impulse response (IIR) filter. The Butterworth type filter was chosen because there is no
ripple which is compliant with the Watterson model. A fourth order filter was chosen for the extremely
narrow transition band. In fact, the transition band is noticeably better than the transition band on some
hardware channel simulators.
In general, the Butterworth IIR filter is easily realized in C. The code and corresponding diagram is
as follows:

```
    /* Now run the noise through a fourth order digital filter */
Fn[0] = value;
Gn[0] = Fn[4]*B[5] + Fn[3]*B[4] + Fn[2]*B[3] + Fn[1]*B[2] + Fn[0]*B[1];
Gn[0] = Gn[0] - Gn[1]*A[2] - Gn[2]*A[3] - Gn[3]*A[4] - Gn[4]*A[5];

Fn[4] = Fn[3];
Fn[3] = Fn[2];
Fn[2] = Fn[1];
Fn[1] = Fn[0];

Gn[4] = Gn[3];
Gn[3] = Gn[2];
Gn[2] = Gn[1];
Gn[1] = Gn[0];
```

For a filter of order n, there are 2*n+1 delays. Therefore the fourth order LPF has a filter startup period, or
initialization period, of 9 samples. For each sample the input to the filter is Gaussian noise generated as
previously described, and the output of the filter is band limited Gaussian noise. The values in the A[n+1]
and B[n+1] arrays define the filter and are calculated at the beginning of ALECALL based on the given

bandwidth.
     In addition to band limiting the Gaussian noise, the amplitude must be adjusted to generate the
predefined SNR ratio. This is a two step process. Before the digital filter, the Gaussian noise is adjusted in
amplitude to match the predefined SNR ratio. This is a simple calculation because the signal power and
SNR ratio are known(noise power = signal power / [SNR ratio]). Unfortunately, the digital filter introduces
power loss. Therefore, the amplitude must be increased after the digital filter to compensate for the power
loss. When calculating the power loss, a good approximation to the 4th order low pass Butterworth filter is
an ideal low pass filter. With this approximation, the power loss is exactly inversely proportional to the
filtered bandwidth. The lost power is accounted for in the following code:

```
    Scale_factor = sqrt( Nyquist / Bandwidth );

    /*
    ** Now, the noise power has been attenuated by the filter.
    ** Multiply by the sqrt( filter_bandwidth / nyquist_frequency) so
    ** that the noise power ( power is average squared value ) is
    ** correct for the given signal to noise ratio.
    */
    Gn[0] = Gn[0] * Scale_factor;
```

Multipath:

     The method used to create multipath is nearly identical to that of Eric Johnson's channel
simulation software. The method implements a simple delay line where the current sample value is stored
and added to a previous sample value an integer number of samples back corresponding to the multipath
delay in milliseconds. Eric Johnson's software implements linear interpolation between sample values for
fractional sample delays. ALECALL is limited to an integer number of sample delays. This provides a
sufficient resolution of 1/44100 seconds or about 0.023 ms. The digital delay line is implemented as
follows:

```
     int multipath( int current )
     {
         int temp;
         Multipath_d_l[ Multi_index++ ] = current;
         if( Multi_index > 441 )                       /* Wrap around */
             Multi_index = 1;
         temp = Multipath_d_l[ wrap(Multi_index - (int)(44.1*Multi_delay) )];
         return (int)( Multi_scale*temp );
     } /* end multipath */
```

The variable Multi_scale is included to allow the multipath signal to have either more or less power than
the original signal. The default value of Multi_scale is 1 representing equal power.

Fading:

     Digital implementation of a fading bandwidth digitally proved to be quite challenging. ALECALL
produces fading analogous to the method used by the Harris RF-3460 channel simulator. The Harris RF-
3460 multiplies two versions of the baseband input signal that differ by 90o by two independent tap gain
multipliers. The independent tap gain multipliers are Gaussian processes which have precisely controlled
power spectrums which define the fading bandwidth. The two versions of the baseband signal are then
combined to produce the fading output. By studying the output of the Harris RF-3460, it is evident that the
levels of the independent tap gain multipliers must be such that the magnitude of the fading output can be
a maximum of twice the amplitude of the original signal. Additionally, it is evident that the average
magnitude of the fading output is equal to the average magnitude of the original signal.

To control the power spectrum, a second order Butterworth IIR low pass filter (LPF) is used. The Watterson model requires the envelope of the digital filter power spectrum to be Gaussian. However, as Eric Johnson discovered, a second order digital low pass filter is a good enough approximation. The fading bandwidth ranges between 0.1 and 10 Hz. Therefore, the cutoff frequency of the low pass filter must range between 0.1 and 10 Hz. These are very low frequencies when compared to the Nyquist frequency of 22.05 kHz.

The input to the LPF must be a Gaussian process as dictated by the Watterson model. For numerical convenience, the input to the LPF is Gaussian noise with values that range between -1 and 1 where 1 represents the largest possible tap gain multiplier value, 0.5 represents unity gain such that the average magnitude of the fading output is equal to the magnitude of the original signal, and 0 represents zero amplitude. By dividing the output of the Gaussian random number generating function by 6, the output is Gaussian noise that ranges from -1 to 1 with a standard deviation of 0.1667. However, the average absolute value is approximately 0.1326 instead of the desired average absolute value of 0.5. ALECALL automatically adjusts the input level to match the desired above criteria. The code to create the input to the fading filter is below:

```
double fix_average = 3.77;   /* fix_average is the number that input*/
                             /* needs to be multiplied by so that the   */
                             /* average absolute value of input is      */
                             /* exactly .5                              */
                             /* .1326 * 3.77 = .5                       */
 input = random()/6;         /* Gaussian random number between -1 and 1 */
 input = fix_average*input;  /* So the average absolute value is .5     */
```

This amplification allows input values to be greater than 1 and less than -1. However, once the input is put through the LPF, this happens less than 0.01% of the time, and when it does it is limited to the range from -1 to 1.

The second order LPF controls the spectrum but has a huge power loss. The power loss is proportional to the area under the magnitude versus frequency curve of the LPF. By normalizing the LPF to have an area of 1 (normalized by scaling the frequency axis by the Nyquist frequency so the value at the Nyquist frequency is 1), the LPF will not introduce power loss. This process involves first computing the area under the LPF, then scaling it so the area is unity. The area can be computed several ways, the best of which is by calculating the area under a Gaussian curve for the given LPF cutoff frequency. ALECALL uses an easily integrated function which is an alternative way to approximate the area under the LPF curve. The code to scale the area under the LPF curves is as follows:

```
     normalized = wa/PI;           /* So normalized varies from 0 to 1  */
                                   /* depending on the cutoff frequency */
     Area_gain = 1 / (normalized * atan(1/normalized) );
     Area_gain = sqrt(Area_gain);
     ...
     Fade_gain_1 = Area_gain*(Gn_f1[0]);
```

Normally, the 90o phase shift is a complicated process involving the Hilbert transform. However, because the ALE tones are deterministic waveforms, it is trivial to introduce a 90o phase shift. When ALECALL is generating the ALE tones, the function is Amplitude*cos(wt). To obtain a waveform that is shifted by 90o, the expression is simply Amplitude*cos(wt +ã/2) or Amplitude*sin(-wt). Both versions of the baseband input signal are multiplied by the two independent tap gain multipliers as shown in the following code:

```
     dac_level[chan] = Levels * Level_scale * cos(w * t); /* Original tone */
      ...
```

```
      if( (Fade) && (chan !=RIGHT) )
      {
          fade_filter_1();                      /* Update Fade_gain_1 */
          fade_filter_2();                      /* Update Fade_gain_2 */

          dac_level[chan] = Fade_gain_1 * dac_level[chan];
          dac_level[chan] +=  Fade_gain_2*Levels*Level_scale*( -sin(w*t) );
      }
```

When the two baseband input signals are combined, this results in a fading output.

Conclusion:

ALECALL is currently capable of simulating the Watterson model channel conditions that have
been used for the FED-STD-1045A testing. However, in testing done in the ITS laboratory, the degraded
tone performance tests did not track directly to tests performed with the Harris RF-3460 channel
simulator. In particular, CCIR Good performance using the degraded tone CD was significantly less than
when using the channel simulator.

Regards Charles G4GUO

18. Re: [HFSIG:806] Re: DSP based sound cards

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Wed, 10 Jan 1996 16:22:41 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960110151926.2057A-100000@ucs.orst.edu>
Rendered body type: text/plain

Hi Charles,

On Wed, 10 Jan 1996, Charles Brain wrote:

> >A further comment on sound cards.
> >
> >It's certainly true that few (if any) of the common 16-bit cards come
> >anywhere near the CD-quality audio that they imply. But for the
> >purpose of hooking up to a HF transceiver, even the worst card is plenty
> >good enough. No reason to spend lots of money here.
> >
> >Phil
> Yes you are right,
>
> What is more important is how accurate the Xtal Oscillator is especially
> on synchronous bit streams aka the message Johan posted a few months
> ago.
>
> I tried to be clever and got my 8ary modem to constantly hunt for bit sync
> rather than getting sync and locking to it. What happen? Well as soon as the
> signal faded it re-synced on the noise and started adding extra bits to the
> bit stream, this really screwed up the ECC !
>
> Regards Charles.
>
>

It took a long time devising and doing on-the-air tests to find a good bit synch algorithm for my TOR programs. Here is one way that I do it (and is due to Peter, G3PLX) that may be of interest:

1) oversample the output from the integegrate-and-dump filters, in my case I use 10 times oversampling - just because it was convenient.

2) what you want to establish is a sliding window that is one symbol long. For the case in 1) the window has ten "ticks" labelled -5, -4, -3, -2, -1, 1, 2, 3, 4, 5. When there is perfect bit synch, there would be a bit transition around tick -1, or 1.

3) the algorithm thus looks at the output of the demodulator and sums the value assigned to the tick where it sees a transition, i.e., if it sees a transition at -3, it adds -3 into a global counter called say, BITSYNC.

4) decide on some thershold value, say +/-128 or so, that when that is exceeded, you need to bump your master clock timing (either change it's

frequency, let it miss a beat/bump it up, or dither - whatever means
that will do it). If the value in BITSYNC is negative for instance, your clock
is too fast, and if it is positive, it is too slow.

The neat thing is that, depending on the noise present, random
transitions, tend to cancel each other in BITSYNCH, so you seldom see
spurious clock adjustments. A nice systematic clock, however, tracks very
nicely. The trick is to find suitable thresholds, however, even that is
not overly critical. For SITOR, this early/late gate procedure tracks
extremely well. The other thing is that since the algorithm communicates
the value of the bit at its mid value, if for example your demodulator
filters delay are 1/2 the symbol time, you effectively achieve near
coherent (true matched filter) performance.

I don't think this algorithm is all that unique and probably find it used
in other situations.

Another very neat way of doing it right, is to do what Dave Mills have
implemented for his RTTY demodulator for the DSP-93. He has made a
Viterbi decoder based on the where the transitions occur and thus
effectively has a ML approach. It is possible to feed the data from each
of the above data points into a trellis and come out with a ML solution
of which instance was the correct one.

An erratic synch algorithm can really give one crazy.

Hope this is of interest.

73's

--Johan

19. Re: [HFSIG:808] Re: DSP based sound cards

From: chbrain@dircon.co.uk (Charles Brain)
Date: Thu, 11 Jan 1996 18:20:47 +0000
To: hfsig@tapr.org
Message-ID: <199601111843.AA05933@felix.dircon.co.uk>
Rendered body type: text/plain

>An erratic synch algorithm can really give one crazy.
>
>Hope this is of interest.
>
>73's
>
>--Johan
>
>
Thanks Johan,
 The technique I used for bit sync was to use MDS as described by Adrian
Nash G4ZHZ. I have arranged the FFT's input window to be one bit period, on
each sample I do an FFT, then take the magnitude of the bin with the
greatest value and divide it by the sum of all the other magnitudes. This
produces a signal with a component at the symbol. I then filter this
and push it though a buffer who's length is half a bit period.

I then look at the current filtered value and the oldest one. If they are above
a certain level and the sample in the middle is larger then I consider I have
straddled the symbol and I find the bin with the greatest magnitude.
I Then lock an NCO to this etc etc etc. The NCO is required as sometimes you
get two symbols with the same tone.

I use the bit sync status as a DCD as well.
I guess I ought to let it continuosly sync but increase the threshold levels
so if the receive signal drifts I can track it. If the signal fades then I
should no longer try to lock onto the noise!

I guess I will have hours of fun perfecting this ! After all thats what it's all
about.

Regards Charles.

20. Re: [HFSIG:809] Re: DSP based sound cards

From: chbrain@dircon.co.uk (Charles Brain)
Date: Thu, 11 Jan 1996 19:25:48 +0000
To: hfsig@tapr.org
Message-ID: <199601111948.AA10625@felix.dircon.co.uk>
Rendered body type: text/plain

The technique I used for bit sync was to use MDS as described by Adrian
>Nash G4ZHZ. I have arranged the FFT's input window to be one bit period, on
^^^^^^^

Whoops I mean't symbol not bit !!!

CHB

21. SS, Digital Comms, and DSP Books - a bibliography

From: srbible@mindport.net (Steven R. Bible)
Date: Thu, 11 Jan 1996 18:09:52 -0500
To: ss@tapr.org, hfsig@tapr.org
Message-ID: <199601112309.SAA16246@polaris.mindport.net>
Rendered body type: text/plain

Recently I was looking for textbooks on the topics of Spread Spectrum,
Digital Communications, and Digital Signal Processing. I turned to the
World Wide Web in search for these and technical book stores advertising on
the Web. Here's what I found.

    http://www.mindport.net/~srbible/biblio.html

The books listed on this page are recent, in-print books.

Hyperlinks on titles will take you to the publishers web page that will
describe the books in more detail. At the end of the book list are some book
stores that specialize in technical books and publishers.

Let me know how you like this format. Let me know if there's a book to add
to the list. Perhaps what would be even more useful to hams is a short
description by a ham on the fitness of the book toward our work. I would
like to rate the books either by stars (1 to 5 stars) or catogorize them as
theoretical, practical, recommended reading, mandatory read, a most have.

Other bibliograph web pages I have authored are:

  Spread Spectrum - http://www.tapr.org/ss/biblio.html

  Packet Radio - http://www.tapr.org/tapr/html/biblio.html

 - Steve, N7HPR

srbible@mindport.net
n7hpr@amsat.org
n7hpr@tapr.org

22. HF Modem sounds uploaded

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Sun, 14 Jan 1996 17:32:13 -0800
To: HFSIG@tapr.org
Message-ID: <9601150126.AA22634@ucs.orst.edu>
Rendered body type: text/plain

Hi,

I have made two .WAV format recodings of Pawel's new modems: the 15-tone
parallel modem, and also of the 30 BPS slow speed BPSK modems - for the
those that does not have an EVM. These sounds may be played on Windows
system using your media player. The sample rate is 22 kHz and the clips are
some 20 seconds each.

Please look for them in a file "SOUNDS.ZIP" on the HFSIG upload area. Please
be warned that the files are big.

--Johan

23. HF channel simulator

From: "Bergzen Hakan, KARL" <Hakan.Bergzen@telub.se>
Date: Mon, 15 Jan 96 11:03:00 GMT
To: HFSIG at TAPR <hfsig@tapr.org>
Message-ID: <31053209@noak>
Rendered body type: text/plain

Hello to you all,

Now with all this exciting modem development going on, wouldn't
it be interesting if an HF channel simulator, preferrably operating

in accordance with the Watterson model (CCIR Rep 549-2). CCIR
(or ITU-R as it is nowadays is called) have also defined 3 typical
channels: Good, Moderate and Poor (CCIR Rec 520-1) beside the
AWGN specifically for the purpose of performing comparative
evaluations on HF modems.

To my knowledge, the available commercial channel simulators
are quite expensive (They can even cost up to USD30000, due to a
small market and small series). They are therefore nothing for the
private developer to consider. However, they are extremely useful
during HF modem development and testing.

The Watterson model is a fairly simple one, easily incorporated in
a cheap fixed point DSP. My suggestion/question is: Has anyone out
there developed a Watterson simulator operating on any of those handy
and available DSP starter kits from Texas Instruments, Motorola...?
And, if so, could that software code be made available (shareware/
freeware) to us all?

All the best,

Hakan Bergzen (hakan.bergzen@telub.se)

24. Re: [HFSIG:813] HF channel simulator

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Mon, 15 Jan 1996 11:51:03 +0100 (MEZ)
To: hfsig@tapr.org
Message-ID: <Pine.A32.3.91.960115114412.21491A-100000@hades.uet.e-technik.th-darmstadt.de>
Rendered body type: text/plain

On Mon, 15 Jan 1996, Bergzen Hakan, KARL wrote:

> Hello to you all,
>
> Now with all this exciting modem development going on, wouldn't
> it be interesting if an HF channel simulator, preferrably operating
> in accordance with the Watterson model (CCIR Rep 549-2). CCIR
> (or ITU-R as it is nowadays is called) have also defined 3 typical
> channels: Good, Moderate and Poor (CCIR Rec 520-1) beside the
> AWGN specifically for the purpose of performing comparative
> evaluations on HF modems.
>
> To my knowledge, the available commercial channel simulators
> are quite expensive (They can even cost up to USD30000, due to a
> small market and small series). They are therefore nothing for the
> private developer to consider. However, they are extremely useful
> during HF modem development and testing.
>
> The Watterson model is a fairly simple one, easily incorporated in
> a cheap fixed point DSP. My suggestion/question is: Has anyone out
> there developed a Watterson simulator operating on any of those handy
> and available DSP starter kits from Texas Instruments, Motorola...?
> And, if so, could that software code be made available (shareware/
> freeware) to us all?
>
> All the best,
>
> Hakan Bergzen (hakan.bergzen@telub.se)
>
>

Hi!

I've here the Watterson simulator running on a TMS320c26 board.
It should be easy to port to the EVM. The problem is: I don't have the
EVM. I wanted to put the source on the the ftp-server, but there are
still some changes to be done and the documentation is currently in
German.

The C26 board I'm using here for the simulator uses an 320c44 AIC, as
far as I know the EVM uses the same chip.

If anybody is interested in porting the simulator to the EVM, please
send me an e-mail.

Three multipath components can be simulated plus AWGN.
There seems to be some computational power left to implement an interference

model (bandpass filtered Gaussian noise).

73' Alexander DL8AAU

----------------------------------+--------------------------------------
Alexander F. Kurpiers	
Institut f. Uebertragungstechnik	Voice: +49-6151-162369
u. Elektroakustik	Fax : +49-6151-165545
Merckstrasse 25	EMail: a.kurpiers@uet.th-darmstadt.de
D-64283 Darmstadt (Germany)	Hamradio: dl8aau@db0zdf.#rpl.deu.eu
----------------------------------+--------------------------------------

25. Re: [HFSIG:814] Re: HF channel simulator

From: chbrain@dircon.co.uk (Charles Brain)
Date: Mon, 15 Jan 1996 18:01:30 +0000
To: hfsig@tapr.org
Message-ID: <199601151841.AA00029@felix.dircon.co.uk>
Rendered body type: text/plain

>The C26 board I'm using here for the simulator uses an 320c44 AIC, as
>far as I know the EVM uses the same chip.
>
>If anybody is interested in porting the simulator to the EVM, please
>send me an e-mail.
>
>73' Alexander DL8AAU
>
>
Hello Alexander,
I would be interested in the code, however I can't understand German
and I have a C50 DSK, it should be possible to do a direct port as
the C50 can be put into C25/6 emulation mode.

Regards Charles G4GUO

26. Near Vertical Angle propigation

From: FLundgren@daniel.maf.org (Fred Lundgren)
Date: Wed, 24 Jan 96 11:03:37 UTC
To: hfsig@tapr.org
Message-ID: <9600248225.AA822507011@gabriel.maf.org>
Rendered body type: text/plain

First of all, thanks to the many who responded to my previous inquire
     on "Radio Mail using Paktor/GTor." There were many useful suggestions.

     The next question I have concerns "Near Vertical Angle propagation"
     (NVA). I have seen NVA mentioned in articles about subsurface antennas
     but, I have been unable to locate data in the literature (that I have
     which isn't all that much) that I can hang my hat on.

     So: Assuming an operating frequency somewhere between 3 and 7 MHZ and a
         desired earth return point around 100 to 200 miles out:

          1) What type of antenna is best? Dipole, Rhombic, etc.

          2) Optimum height above ground?

          3) Best ground? earth, metal, counterpoise. etc.

          4) Optimum takeoff angle and how to optimize?

          5) What to expect in the way of power loss to ground, vs height
                above ground, vs takeoff angle etc.

          6) What to expect in the way of SWR and overall antenna
                efficiency in an optimum NVA system?

     Thanks for the assist on this.

     73, Fred (N7LSZ)

27. Re: [HFSIG:816] Near Vertical Angle propigation

From: Alan Bloom <alanb@tsnake2.sr.hp.com>
Date: Wed, 24 Jan 1996 12:12:04 -0800 (PST)
To: hfsig@tapr.org (Fred Lundgren)

Message-ID: <199601242012.AA168054324@tsnake2.sr.hp.com>
Rendered body type: text/plain

> The next question I have concerns "Near Vertical Angle propagation"
> (NVA).

>From what I gather, a "NVA" aka "NVIS" antenna is what we used to call a "PW" antenna ("puny weak"). A good DX antenna has good low-angle radiation to maximize long-distance propagation and the least possible high-angle radiation to minimize interference from nearby stations and noise sources. A NVIS ("Near-Vertical Incident Skywave") antenna does just the opposite, in order to facilitate local (under a couple hundred miles) communications.

> So: Assuming an operating frequency somewhere between 3 and 7 MHZ and a
> desired earth return point around 100 to 200 miles out:
>
> 1) What type of antenna is best? Dipole, Rhombic, etc.

I don't think it matters much as long as the antenna is close to the ground.

> 2) Optimum height above ground?

The closer to the ground, the less low-angle radiation. But if you get too close, you lose too much efficiency due to ground losses. A quarter wavelength seems like a good height since the ground-reflected wave is in phase (maximum radiation) for vertical radiation.

> 3) Best ground? earth, metal, counterpoise. etc.

If you could cover the ground under the antenna with wire mesh, that would minimize ground losses.

> 4) Optimum takeoff angle and how to optimize?

You want to maximize high-angle radiation and minimize low-angle.

> 5) What to expect in the way of power loss to ground, vs height
> above ground, vs takeoff angle etc.

It depends so much on your particular soil conditions that it's impossible to give a general answer. There are computer programs available that give accurate answers if you can somehow figure out the conductivity and dielectric constant of your soil.

> 6) What to expect in the way of SWR and overall antenna
> efficiency in an optimum NVA system?

Antenna impedance (at least for simple antennas like dipoles) gets lower the closer the antenna is to ground. Again, the efficiency (and SWR) depend strongly on your soil conditions. For maximum efficiency, you want the soil conductivity to be high (good reflector).

Alan Bloom N1AL

28. Re:Near Vertical Angle propigation

From: Rick Whiting <rwhiting@winternet.com>
Date: Wed, 24 Jan 1996 14:59:56 -0600
To: hfsig@tapr.org
Message-ID: <199601242059.OAA05366@icicle>
Rendered body type: text/plain

At 12:44 1/24/96 -0600, Fred Lundgren wrote:
[snip]
>
> The next question I have concerns "Near Vertical Angle propagation"
> (NVA). I have seen NVA mentioned in articles about subsurface antennas
> but, I have been unable to locate data in the literature (that I have
> which isn't all that much) that I can hang my hat on.
>
> So: Assuming an operating frequency somewhere between 3 and 7 MHZ and a
> desired earth return point around 100 to 200 miles out:
>
> 1) What type of antenna is best? Dipole, Rhombic, etc.

Dipole is one. See reference below for more.

>
> 2) Optimum height above ground?

Low. You can model this with appropriate software, e.g., see ads in ham magazines. Be sure to pick a package that handles ground correctly (MININEC doesn't). Also see info in refeence below.

Take a look at "NVIS (Near Vertical Incident Skywave), Running Reference File," from Stanly E. Harter, Assistant Chief, Telecommunications Div., Office of the Governor, Office of Emergency Services, 2800 Meadowview Rd., Sacramento, CA 95832. Phone (916) 427-4281 (or try packet to KH6GBX @ WA6NWE.#NOCAL.CA.USA). This is a collection of several articles that will answer many of your questions.

73/Rick W0TN

Richard A. (Rick) Whiting	Phone: + 1 612 550 1213
5780 Rosewood Ln. N.	E-mail: rwhiting@winternet.com
Plymouth, MN 55442-1411	Packet: W0TN @ WB0GDB.MN.USA.NOAM
U.S.A.	Fax: Number on request

29. Re: PLEASE HELP: Radio Design for Severely Handicapped

From: Rick Whiting <rwhiting@winternet.com>
Date: Wed, 24 Jan 1996 15:30:14 -0600
To: hfsig@tapr.org
Message-ID: <199601242130.PAA10989@icicle>
Rendered body type: text/plain

Seth A. Watkins wrote:

>We are interested in designing/acquiring/modifying a radio (digital?) for
>use by a severely handicapped individual who was a student before a
>terrible accident. Only breath-control is practical in this situation.
>Control of power and tuning are obviously the essential functions needed.
[snip]

Seth:

I recommend you contact the HandiHams at the Courage Center in Minneapolis. They have a long history of using radio with handicapped people. The address is: Courage Center, 3915 Golden Valley Rd., Golden Valley, MN 55422. Phone: (612) 520-0520.

Regards/ Rick W0TN

Richard A. (Rick) Whiting	Phone: + 1 612 550 1213
5780 Rosewood Ln. N.	E-mail: rwhiting@winternet.com
Plymouth, MN 55442-1411	Packet: W0TN @ WB0GDB.MN.USA.NOAM
U.S.A.	Fax: Number on request

30. RE: [HFSIG:816] Near Vertical Angle propigation

From: "Sadowski,P O4 USAF JCSAR/EDD" <sadowskp@jcsar.nellis.af.mil>
Date: Thu, 25 Jan 96 07:42:00 cst
To: hfsig <hfsig@tapr.org>
Message-ID: <31078BC8@mailgtwy.nellis.af.mil>
Rendered body type: text/plain

I wish I could remember which issue of the AFCEA (Armed Forces Communications Electronics Association) Journal I read the article in... but somewhere between five and three years ago there was an article on how the military uses this mode. You can probably get a copy of it somewhere on the Internet... or via AFCEA.. or a really good library...

Bottom line... because of the need to remain mobile... the typical situation is to use the extra long vertical whip... and bend it over horizontal... admittedly... the preferred frequencies for that use are the SINCGARS freqs/radio... mostly from 30MHz to 70MHz I believe...

Hope this helps....

>From: hfsig
>To: hfsig
>Subject: [HFSIG:816] Near Vertical Angle propigation
>Date: Wednesday, January 24, 1996 12:38PM

> First of all, thanks to the many who responded to my previous inquire
> on "Radio Mail using Paktor/GTor." There were many useful suggestions.

> The next question I have concerns "Near Vertical Angle propagation"
> (NVA). I have seen NVA mentioned in articles about subsurface antennas

> but, I have been unable to locate data in the literature (that I have
> which isn't all that much) that I can hang my hat on.

> So: Assuming an operating frequency somewhere between 3 and 7 MHZ and
a

> desired earth return point around 100 to 200 miles out:

> 1) What type of antenna is best? Dipole, Rhombic, etc.

> 2) Optimum height above ground?

> 3) Best ground? earth, metal, counterpoise. etc.

> 4) Optimum takeoff angle and how to optimize?

> 5) What to expect in the way of power loss to ground, vs height

> above ground, vs takeoff angle etc.

> 6) What to expect in the way of SWR and overall antenna

> efficiency in an optimum NVA system?

> Thanks for the assist on this.

> 73, Fred (N7LSZ)

31. Re: [HFSIG:816] Near Vertical Angle propigation

From: k5yfw@sacdm01.kelly.af.mil (WALT DUBOSE - K5YFW)
Date: Thu, 25 Jan 96 10:35:13 -0600
To: hfsig@tapr.org
Message-ID: <9601251635.AA15337@sacdm01.kelly.af.mil>
Rendered body type: text/plain

Fred, et al.

In your message of 24 Jan 1996 at 1238 CST, you write:
> The next question I have concerns "Near Vertical Angle propagation"
> (NVA). I have seen NVA mentioned in articles about subsurface antennas
> but, I have been unable to locate data in the literature (that I have
> which isn't all that much) that I can hang my hat on.

I'm not familiar with NVA but with NVIS (near vertical
incident skywave) antenna.

>
> So: Assuming an operating frequency somewhere between 3 and 7 MHZ and a
> desired earth return point around 100 to 200 miles out:
>
> 1) What type of antenna is best? Dipole, Rhombic, etc.

A dipole mounted less than 1/4 wavelength above ground (for
40 meters, less than 25 ft in most U.S. locals) will produce
a NVIS antenna pattern. Something around 40-50 ft will do
the same thing for 80 meters.

A Rhombic generally is considered a low radiation angle
antenna...good for skip/DX work. I have however, seen a
rhombic pointed upward but couldn't say what kind of antenna
pattern you would get and if it would work for 3-7 MHz for
close in...100-200 miles.

>
> 2) Optimum height above ground?

See Above
>
> 3) Best ground? earth, metal, counterpoise. etc.

Ah ha...that's a good question. In Saudi, we used much
chain-link fence, chicken-wire and aluminium sheets to create
artificial grounds. We found that chicken-wire worked as
well as anything else. We mounted the 8 Mhz and 11 Mhz
dipoles 3/8 wavelength above the chicken-wire ground. From

the center of Saudi, we were able to work the entire length
and breath of the country during the day...an far into the
night. Usually changing to 3-5 Mhz channels after mid-night.~

Look at a good antenna book that has a U.S. ground
conductance map in it and use a propagation program like
ELNEC, MiniNec to get a plot of your antenna pattern.

>
>   4) Optimum takeoff angle and how to optimize?

That's going to depend on the condition of the ionosphere.
Generally you want radiation above 45 deg for NVIS operation.

>
>   5) What to expect in the way of power loss to ground, vs height
>   above ground, vs takeoff angle etc.

Gee, the idea is to get as much signal above 45 deg vertical.
unless you're doing a real fixed site, emergency
communications, etc, use a program something like ELNEC to
help you. If this is a commercial site...give me a landline
and I refer you to a commercial firm that does this sort of
work.

>
>   6) What to expect in the way of SWR and overall antenna
>   efficiency in an optimum NVA system?

For a 1/2 wavelength dipole mounted 3/8 wavelength above
ground, the SWR should be less than 2:1 (between 45-55 ohms
impedance).

Hope this helps.

73, Walt/K5YFW

32. Chirp Waveforms

From: chbrain@dircon.co.uk (Charles Brain)
Date: Thu, 25 Jan 1996 19:25:45 +0000
To: hfsig@tapr.org
Message-ID: <199601251951.AA18973@felix.dircon.co.uk>
Rendered body type: text/plain

Does anyone have any literature on H.F chirp modems, I looked in my
copy of the ARRL Spread spectrum Source Book, but the only thing I found
apart from a brief mention of chirp as used in radars was abot DS SS.

I was thinking of doing some emulations of a simple chirp waveform, i.e
a linear chirp from say 500Hz to 2700Hz at a 10ms repetion rate. I can
emulate the waveform in PC-DSP.
 With an increase in freq respresenting logic 1 and a decrease logic 0. Then
using a matched filter for decoding it. This would give an uncoded data rate
of 100 baud.

On top of that I could use Phil's Viterbi stuff to provide ECC.

I want something that has the potential of providing a slow robust waveform
for use possibly on 3.5 Mhz at night under severe multi-path conditions.

Question is a linear chirp the ideal waveform, none of the books I have cover
this topic.

Regards Charles G4GUO

33. Re: [HFSIG:820] RE: Near Vertical Angle propigation

From: k5yfw@sacdm01.kelly.af.mil (WALT DUBOSE - K5YFW)
Date: Thu, 25 Jan 96 14:35:50 -0600
To: hfsig@tapr.org
Message-ID: <9601252036.AA21287@sacdm01.kelly.af.mil>
Rendered body type: text/plain

In your message of 25 Jan 1996 at 0953 CST, you write:
>
> I wish I could remember which issue of the AFCEA (Armed
> Forces Communications Electronics Association) Journal
> I read the article in... but somewhere between five and three
> years ago there was an article on how the military uses this

> mode. You can probably get a copy of it somewhere on the
> Internet... or via AFCEA.. or a really good library...

I can't remember either but I do know that most/much of the article / research was done by Chris Christensin (?) W0TXX who resides in Bellville, IL, used to work, at Scott AFB and was chief of AF MARS for a while. Now he has a consulting firm supporting Military HF communications.

BTW, Chris has a couple of books for sale on Tactical HF Comm.

>
> Bottom line... because of the need to remain mobile... the typical
> situation is to use the extra long vertical whip... and bend it over
> horizontal... admittedly... the preferred frequencies for that use
> are the SINCGARS freqs/radio... mostly from 30MHz to 70MHz
> I believe...
>

The Marines and Army needed communications beyond the 25-30 miles that LowBand VHF or HF groundwave could provide. An HF whip (16ft) provided good communications on groundwave and 700 or so miles and out. Bending the antenna (whip) over and putting a loading coil in the middle helped the Marines (In Saudi and later the Army) achieve HF coverage from 30-500/700 miles with their mobiles (HumVees). This all came about through an AFCEA course that Chris taught in Virginia in about '88 or '89.

73, Walt/K5YFW

> Hope this helps....
>
> ----------
> >From: hfsig
> >To: hfsig
> >Subject: [HFSIG:816] Near Vertical Angle propigation
> >Date: Wednesday, January 24, 1996 12:38PM
>
> >   First of all, thanks to the many who responded to my previous inquire
> >   on "Radio Mail using Paktor/GTor." There were many useful suggestions.
>
> >   The next question I have concerns "Near Vertical Angle propagation"
> >   (NVA). I have seen NVA mentioned in articles about subsurface antennas
>
> >   but, I have been unable to locate data in the literature (that I have
> >   which isn't all that much) that I can hang my hat on.
>
> >   So: Assuming an operating frequency somewhere between 3 and 7 MHZ and
> a
>
> >   desired earth return point around 100 to 200 miles out:
>
> >   1) What type of antenna is best? Dipole, Rhombic, etc.
>
> >   2) Optimum height above ground?
>
> >   3) Best ground? earth, metal, counterpoise. etc.
>
> >   4) Optimum takeoff angle and how to optimize?
>
> >   5) What to expect in the way of power loss to ground, vs height
>
> >   above ground, vs takeoff angle etc.
>
> >   6) What to expect in the way of SWR and overall antenna
>
> >   efficiency in an optimum NVA system?
>
> >   Thanks for the assist on this.
>
> >   73, Fred (N7LSZ)
>

34. RE: [HFSIG:822] Chirp Waveforms

From: "Sadowski,P O4 USAF JCSAR/EDD" <sadowskp@jcsar.nellis.af.mil>
Date: Thu, 25 Jan 96 13:55:00 cst
To: hfsig <hfsig@tapr.org>
Message-ID: <3107E3BD@mailgtwy.nellis.af.mil>
Rendered body type: text/plain

Out of curiosity, what would be frequency increment at each 10 ms interval ?
I think I may lack understanding of what you propose - seems like your
spectral bandwidth would be a good fraction of a MHz ... I did chirp stuff
with matched filters on radars a few years back... a superior technique to
up S/N and get better range resolution at the same time.... impossible
without ALOT more power otherwise....However, I'm not smart enuff to see
how this technique is gonna benefit you for HF data modes.. but I'm
over my head here... I also saw chirped waveforms used in CW illuminator
radars (semi-active homing missiles).... the chirped waveform was to
do a velocity gate pulloff... fun stuff itself... but not a HF data
waveform...
But I'm real curious... and I can learn...

Maybe you are just wanting to do spread spectrum... not a hopper but a
direct sequence (convolution coded) system on HF ??? now that I understand.

Thanks
AH6LS

From: hfsig
To: hfsig
Subject: [HFSIG:822] Chirp Waveforms
Date: Thursday, January 25, 1996 1:58PM

Does anyone have any literature on H.F chirp modems, I looked in my
copy of the ARRL Spread spectrum Source Book, but the only thing I found
apart from a brief mention of chirp as used in radars was abot DS SS.

I was thinking of doing some emulations of a simple chirp waveform, i.e
a linear chirp from say 500Hz to 2700Hz at a 10ms repetion rate. I can
emulate the waveform in PC-DSP.
 With an increase in freq respresenting logic 1 and a decrease logic 0. Then
using a matched filter for decoding it. This would give an uncoded data rate
of 100 baud.

On top of that I could use Phil's Viterbi stuff to provide ECC.

I want something that has the potential of providing a slow robust waveform
for use possibly on 3.5 Mhz at night under severe multi-path conditions.

Question is a linear chirp the ideal waveform, none of the books I have
cover
this topic.

Regards Charles G4GUO

35. Re: [HFSIG:816] Near Vertical Angle propigation

From: gc@fox.cen.com (Gary Chatters)
Date: Thu, 25 Jan 96 17:12:17 EST
To: hfsig@tapr.org
Message-ID: <9601252212.AA03295@cen.com>
Rendered body type: text/plain

> The next question I have concerns "Near Vertical Angle propagation"
> (NVA). I have seen NVA mentioned in articles about subsurface antennas
> but, I have been unable to locate data in the literature (that I have
> which isn't all that much) that I can hang my hat on.
>

A web search produced some interesting results; more than I can
summarize here. The best search string seemed to be "vertical incidence".
Some of the web pages found discussed ionospheric sounding.

The one piece of information that came up but hasn't been mentioned
is the use of loop antennas. The web page that I found was for
DRAE (brand) magnetic loop antennas, but would probably apply to any of the
ones that are available (or, of course, could be homebrewed).

The antenna described was a 1.7 meter diameter loop which could be
tuned over the range 3-10.3 MHz and was described as
"It is particularly suited for daytime
Near Vertical Incidence Short skip propagation on 40 and 80 metres,..."

Since I live in an apartment, I have wanted to try a loop antenna.

This looks like a good excuse for giving one a try this year.

Gary

36. Re: [HFSIG:824] RE: Chirp Waveforms

From: chbrain@dircon.co.uk (Charles Brain)
Date: Thu, 25 Jan 1996 22:27:37 +0000
To: hfsig@tapr.org
Message-ID: <199601252253.AA27818@felix.dircon.co.uk>
Rendered body type: text/plain

>
>Out of curiosity, what would be frequency increment at each 10 ms interval ?
Sorry I obviously didn't explain myself correctly.

the chirp would start at a frequency of 500hz and 10ms later would be at a
frequency of 2700 Hz for logic 1 and for logic 0 would start at 2700Hz and
10ms later would be back at 500hz. Then using an FIR filter matched to the
chirp I would extra the signal (i.e a filter whose impulse response is the flip
of the chirp).

I have done some quick simulations since I posted and the waveform seems to
work well superimposed on a MFSK signal of about 10 db stronger. I will
have to look at the spectrum. Principally I am looking for something that can
tolerate multipath and is immune to CW and RTTY "interference". Ideally it
would be nice to swipe the waveform over a much larger BW.

80meters is virtually unusable for "reliable" communications in Europe during
hours of darkness due mainly to commercial encrypted SSB transmissions
and other things.

Regards Charles.

37. Re: [HFSIG:822] Chirp Waveforms

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Thu, 25 Jan 1996 19:38:50 -0800
To: hfsig@tapr.org
Message-ID: <9601260337.AA31827@ucs.orst.edu>
Rendered body type: text/plain

Charles,

>Does anyone have any literature on H.F chirp modems, I looked in my
>copy of the ARRL Spread spectrum Source Book, but the only thing I found
>apart from a brief mention of chirp as used in radars was abot DS SS.
>

It may be worthwhile to look in the IEE Journal of Communications, Speech
and Vision. Volume 139, Number 4, August 1992. (Not the IEEE, but the IEE,
i.e., the British one). "A chirp modem incorporating interference excision"
by Darbyshire and Gott.

The objective is a 75 baud modem for HF that has special considerations for
adjacent channel interference and tolerance to multipath affects.

In this one, DPSK chirping is done using 2.7 kHz BW - hence one may consider
it a form of SS. A nice readable paper with plenty of stimulating thoughts.

--Johan

38. Re: [HFSIG:826] RE: Chirp Waveforms

From: erik@ve7mdl.ampr.org
Date: Thu, 25 Jan 96 22:39:41 PST
To: hfsig@tapr.org
Message-ID: <24024@ve7mdl.ampr.org>
Rendered body type: text/plain

Charles,

Have you looked at Clover? It seems to perform very well in poor HF
conditions. The Hal Communications P38 card gives you up to 400 bps,
but the speed depends on the conditions. It adaps automatically.

I have also found it more persistent than other HF digital modes. It
keeps trying to keep the link alive, so it can handle rather long fades.

73 de VE7MDL   ....Erik.

39. Re: [HFSIG:828] RE: Chirp Waveforms

From: chbrain@dircon.co.uk (Charles Brain)
Date: Fri, 26 Jan 1996 07:08:17 +0000
To: hfsig@tapr.org
Message-ID: <199601260734.AA15773@felix.dircon.co.uk>
Rendered body type: text/plain

>Charles,
>
>Have you looked at Clover? It seems to perform very well in poor HF
>conditions. The Hal Communications P38 card gives you up to 400 bps,
>but the speed depends on the conditions. It adaps automatically.
>
>I have also found it more persistent than other HF digital modes. It
>keeps trying to keep the link alive, so it can handle rather long fades.
>
>73 de VE7MDL   ....Erik.
>
>
Hello Erik,
 Yes I do have a Clover modem, it is better than normal FSK systems.
However it's not really there yet. I think the article in QEX by KE4BAD from
SHAPE Technical center says it all, i.e comparing it to a STANAG 4285 modem
operating at 75 baud.

-->

Hello Johan,
 I will see if someone at work has that magazine, I am not a member of
that organisation but some of the people I work with are.

Regards Charles

40. Flash boot prog for EVM56002 uploaded

From: "Helmut Strickner" <hstr@gil.com.au>
Date: Fri, 26 Jan 1996 17:53:23 +1000
To: hfsig@tapr.org
Message-ID: <199601260753.CAA09121@iccu6.ipswich.gil.com.au>
Rendered body type: text/plain

Hello,

anyone else had problems with programming the flash PEROM on the
EVM56002 board ? The Motorola supplied flash.asm is buggy and does
not cater for the partitioned memory space configuration properly.

I have modified some code for use with Johan's BOOT.ASM and uploaded
it to the /hfsig/upload directory.

Keep up the good work Johan. Any comments are appreciated.

Helmut Strickner VK4STR

41. Re: [HFSIG:827] Re: Chirp Waveforms

From: Danny@edstks1.demon.co.uk (Danny Higgins)
Date: Fri, 26 Jan 1996 09:13:57 GMT
To: hfsig@tapr.org
Message-ID: <1064@edstks1.demon.co.uk>
Rendered body type: text/plain

Hi Charles.

I have been involved a bit in HF narrowband chirp modems.

Try looking at "Introduction to Radar Systems" by Merrill Skolnik,
pages 422 to 427 for a general introduction.

Professor Geoff Gott from UMIST (he is licensed as G3???) has been
doing work on this subject for about 10 years. He gave a paper on it
at the 5th International Conference on HF Radio System Design in 1991.
The IEE publication number is 339 and the article is on pages 212 to 216.
If you have trouble getting hold of it, I have a copy and can send it to
you.

I've not had any time to play with the EVM5X since we last spoke. In fact
I have not been on the air since October. If you manage to get a chirp
modem running in your EVM5X then we could perhaps do some trials on 80m,
with Clover as a back-up.

Regards,
--
Danny Higgins, G3XVR

42. Re: [HFSIG:822] Chirp Waveforms

From: "Bergzen Hakan, KARL" <Hakan.Bergzen@telub.se>
Date: Fri, 26 Jan 96 11:01:00 GMT
To: HFSIG at TAPR <hfsig@tapr.org>
Message-ID: <3113B262@noak>
Rendered body type: text/plain

Hello Charles,

I gather it is the Gott Chirp modem you are referring to. It has been described
by him in lots of conferences, e.g IEE HF Systems and Techniques (74 up to the 86 conference) and also in the Nordic HF89 conference.

It is the simplicity of coping with narrowband interfernce by inserting notch filters
that is one of its strengths. You split the the incoming signal to pass x number of
bandpass filters connected in parallel and covering the signal bandwidth, if you
have an interernce within one of those segments you just block the output from
that segment. This is much easier accomplished than how it is done within the
serial tone modems. Other strongholds: Utilization of the whole signal bandwidth,
slow symbol modulation and simple DPSK demodulation (thus using multipath constructively, as in some serial tone modems).

We tested this modem several years ago and it did not have the performance expected. This could, however, be partly blamed on the prototype status of the modems we were testing and not on the method as such. I would guess it has potential to perform better than MFSK within a given bandwidth.

For what it's worth...

Hakan (hakan.bergzen@telub.se)

43. Re: [HFSIG:830] Flash boot prog for EVM56002 uploaded

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Fri, 26 Jan 1996 09:14:27 -0800
To: hfsig@tapr.org
Message-ID: <9601261713.AA15682@ucs.orst.edu>
Rendered body type: text/plain

Hello Helmuth,

Thanks a lot for sharing the code.

There also is now an program available that a fellow posted on COMP.DSP that allows one to write an image of the P, X, and Y memories to flash and boot with that for an initialized data memory. It does require a bit of manual adjustments before doing so. I am going to write a C program to interactively set up the the details and write out an asembly program that does the exact code. I'll post my version when its done.

Besides that, I have finally gotten all the parts to make up the new interface and will post a short article on the updated schematics, parts list/numbers and where to get everything. Hopefully we can get started on doing some testing on the air. I have gotten another EVM and computer that will be dedicated for a beacon. For those that may be interested, the latest EVM now sports a 66 MHz DSP chip. Motorola also hints that the chip will do up to 80 MHz but will not guarantee that.

--Johan

>Hello,
>
>anyone else had problems with programming the flash PEROM on the
>EVM56002 board ? The Motorola supplied flash.asm is buggy and does

>not cater for the partitioned memory space configuration properly.
>
>I have modified some code for use with Johan's BOOT.ASM and uploaded
>it to the /hfsig/upload directory.
>
>Keep up the good work Johan. Any comments are appreciated.
>
>Helmut Strickner VK4STR
>
>

44. Re: [HFSIG:832] Re: Chirp Waveforms

From: chbrain@dircon.co.uk (Charles Brain)
Date: Fri, 26 Jan 1996 17:47:43 +0000
To: hfsig@tapr.org
Message-ID: <199601261813.AA18435@felix.dircon.co.uk>
Rendered body type: text/plain

>
>Hello Charles,
>
>I gather it is the Gott Chirp modem you are referring to. It has been
>
>Hakan (hakan.bergzen@telub.se)
>
>
Hello Hakan,
 Glad you got back o.k.

Yes bit embaressing this, the prototype was partialy developed at Baddow
Research and I have actually seen it in the flesh.

The idea I wrote about recently was disregarded by the authors as it
doesn't work too well in a dispersive channels! Still it's an interesting paper.

Regards Charles

45. Re: [HFSIG:834] Re: Chirp Waveforms

From: chbrain@dircon.co.uk (Charles Brain)
Date: Fri, 26 Jan 1996 20:39:53 +0000
To: hfsig@tapr.org
Message-ID: <199601262105.AA25743@felix.dircon.co.uk>
Rendered body type: text/plain

>doesn't work too well in a dispersive channels! Still it's an interesting
paper.
>   ^^^^^^^^^^^^^^^^^^
Woops !

I think I mean't it requires more dispersion than swept DPSK. I just know
enough to be dangerous!

CHB

46. Re: [HFSIG:832] Re: Chirp Waveforms

From: Barry McLarnon VE3JF <bm@lynx.ve3jf.ampr.org>
Date: Sat, 27 Jan 1996 04:03:11 +0000 (GMT)
To: hfsig@tapr.org
Message-ID: <199601270403.EAA06429@lynx.ve3jf.ampr.org>
Rendered body type: text/plain

Here's a couple more references... older, but at least they're not in
obscure publications. :-)

A.J. Berni and W.D. Gregg, "On the utility of chirp modulation for
digital signalling", IEEE Trans. Communications, Vol. COM-21, June 1973,
pp. 748-751.

G.F. Gott and J.P. Newsome, "HF data transmission using chirp signals",
Proc. IEEE, Vol. 118, No. 9, September 1971.

Barry

--
Barry McLarnon VE3JF/VA3TCP
Ottawa Amateur Radio Club Packet Working Group
Email: bm@hydra.carleton.ca or bm@lynx.ve3jf.ampr.org

47. Re: [HFSIG:834] Re: Chirp Waveforms

From: "Daniel R. Emblidge" <emblidge@moran.com>
Date: Tue, 30 Jan 1996 13:03:29 -0500
To: hfsig@tapr.org
Message-ID: <310E5D71.4B01@moran.com>
Rendered body type: text/plain

Charles Brain wrote:
>
> >
> > >Hello Charles,
> >
> > >I gather it is the Gott Chirp modem you are referring to. It has been
> >
> > >Hakan (hakan.bergzen@telub.se)
> >
> >
> > Hello Hakan,
> > Glad you got back o.k.
> >
> > Yes bit embaressing this, the prototype was partialy developed at Baddow
> > Research and I have actually seen it in the flesh.
> >
> > The idea I wrote about recently was disregarded by the authors as it
> > doesn't work too well in a dispersive channels! Still it's an interesting paper.
> >
> > Regards CharlesGET ME OFF THIS THING

48. Re: [HFSIG:835] Re: Chirp Waveforms

From: "Daniel R. Emblidge" <emblidge@moran.com>
Date: Tue, 30 Jan 1996 13:03:58 -0500
To: hfsig@tapr.org
Message-ID: <310E5D8E.6D84@moran.com>
Rendered body type: text/plain

Charles Brain wrote:
>
> > >doesn't work too well in a dispersive channels! Still it's an interesting
> > paper.
> > >   ^^^^^^^^^^^^^^^^^^
> > Woops !
> >
> > I think I mean't it requires more dispersion than swept DPSK. I just know
> > enough to be dangerous!
> >
> > CHBGET ME OFF THIS THING

49. Re: [HFSIG:836] Re: Chirp Waveforms

From: "Daniel R. Emblidge" <emblidge@moran.com>
Date: Tue, 30 Jan 1996 13:04:18 -0500
To: hfsig@tapr.org
Message-ID: <310E5DA2.7056@moran.com>
Rendered body type: text/plain

Barry McLarnon VE3JF wrote:
>
> Here's a couple more references... older, but at least they're not in
> obscure publications. :-)
>
> A.J. Berni and W.D. Gregg, "On the utility of chirp modulation for
> digital signalling", IEEE Trans. Communications, Vol. COM-21, June 1973,
> pp. 748-751.
>
> G.F. Gott and J.P. Newsome, "HF data transmission using chirp signals",
> Proc. IEEE, Vol. 118, No. 9, September 1971.
>
> Barry
>
> --
> Barry McLarnon VE3JF/VA3TCP
> Ottawa Amateur Radio Club Packet Working Group
> Email: bm@hydra.carleton.ca or bm@lynx.ve3jf.ampr.orgGET ME OFF THIS THING

50. Punctured convolutional codes

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Tue, 30 Jan 1996 13:29:25 -0800
To: HFSIG@tapr.org
Message-ID: <11BBAC13DED@frl.orst.edu>

Rendered body type: text/plain

Hi,

I have recently come across the use of so called "punctured" convolutional codes in the MIL-STD-188 modems, and also as used in Pactor II. The effect is to change the coding rate from say 1/2 to 7/8 depending on the quality of the link.

I think I understand the general idea behind how and why it is done, but would appreciate anyone (please Phil) elaborate a bit more about the pro's and con's. A good reference would also help a great deal.

Thanks.

--Johan, KC7WW

51. Re: Chirp Waveforms

From: "Bergzen Hakan, KARL" <Hakan.Bergzen@telub.se>
Date: Tue, 30 Jan 96 22:46:00 GMT
To: HFSIG at TAPR <hfsig@tapr.org>
Message-ID: <31199DCA@noak>
Rendered body type: text/plain

Daniel R. Emblidge wrote:
> ...snip...
>GET ME OFF THIS THING

Sorry, I don't understand this comment (and I seem to be at least partly targeted)! Is this a desperate way of trying to stop subscribing, or is it some sort of self-appointed judgement on what types of modem developments that are allowed to be discussed in this newsgroup? (Bugs attack protection has not been needed so far)

Have a nice day on you too...

Hakan (hakan.bergzen@telub.se)

52. Re: [HFSIG:840] Punctured convolutional codes

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 30 Jan 1996 14:39:03 -0800 (PST)
To: forrerj@frl.orst.edu
Cc: hfsig@tapr.org
Message-ID: <199601302239.OAA17042@servo.qualcomm.com>
Rendered body type: text/plain

Punctured codes are actually quite simple. The sender runs the encoder to generate the encoded symbols, but not all of them are sent. By prearrangement between sender and receiver, only a certain fraction are actually sent over the air.

At the receiver, the unsent symbols are replaced with erasures (the symbol value corresponding to an equal likelihood of a 0 and 1). The decoder then runs normally.

This requires a soft-decision decoder, or at least one that can handle three received symbol values (0, 1 and erased/unknown).

Although puncturing a code to a given rate doesn't perform quite as well as a code designed for that specific rate, punctured codes have the significant advantage of being easily implemented and of being easily varied in code rate; a single encoder and decoder can serve for a wide variety of code rates. They are especially suitable for Type II Hybrid ARQ/FEC schemes where the punctured symbols can be optionally sent in a subsequent transmission if needed by current link conditions.

Phil

53. Re: [HFSIG:841] Re: Chirp Waveforms

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Tue, 30 Jan 1996 15:37:36 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960130153310.11697A-100000@ucs.orst.edu>
Rendered body type: text/plain

Hi Hakan,

I suspect this gentleman is desparate to get unsubscribed. I e-mailed him how to use the TAPR web page to unsubscribe. Somehow he must have gotten

subscribed by surfing the net and accidentally selected the wrong box on
the web page. Actually I think that procedure for allowing folks to
subscribe like that is inviting this kind of problem.

Hopefully he got it fixed.

--Johan

On Tue, 30 Jan 1996, Bergzen Hakan, KARL wrote:

>
> Daniel R. Emblidge wrote:
> > ...snip...
> >GET ME OFF THIS THING
>
> Sorry, I don't understand this comment (and I seem to be at least partly
> targeted)! Is this a desperate way of trying to stop subscribing, or is it
> some sort of self-appointed judgement on what types of modem developments
> that are allowed to be discussed in this newsgroup? (Bugs attack protection
> has not been needed so far)
>
> Have a nice day on you too...
>
> Hakan (hakan.bergzen@telub.se)
>
>

54. Re: [HFSIG:840] Punctured convolutional codes

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Tue, 30 Jan 1996 15:54:39 -0800 (PST)
To: Phil Karn <karn@qualcomm.com>
Cc: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960130153754.11697B-100000@ucs.orst.edu>
Rendered body type: text/plain

Hi Phil,

Excellent!

So I gather all that happens is that the coding gain takes a hit as
one starts puncturing, but assuming one use it only when there are
adequate S/N, that should be OK. Besides the little bit of additional
overhead, sounds like a very clever idea to increase throughput.

--Johan

On Tue, 30 Jan 1996, Phil Karn wrote:

> Punctured codes are actually quite simple. The sender runs the encoder
> to generate the encoded symbols, but not all of them are sent. By
> prearrangement between sender and receiver, only a certain fraction
> are actually sent over the air.
>
> At the receiver, the unsent symbols are replaced with erasures (the
> symbol value corresponding to an equal likelihood of a 0 and 1). The
> decoder then runs normally.
>
> This requires a soft-decision decoder, or at least one that can handle
> three received symbol values (0, 1 and erased/unknown).
>
> Although puncturing a code to a given rate doesn't perform quite as
> well as a code designed for that specific rate, punctured codes have
> the significant advantage of being easily implemented and of being
> easily varied in code rate; a single encoder and decoder can serve for
> a wide variety of code rates. They are especially suitable for Type II
> Hybrid ARQ/FEC schemes where the punctured symbols can be optionally
> sent in a subsequent transmission if needed by current link conditions.
>
> Phil

55. Re: [HFSIG:840] Punctured convolutional codes

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 30 Jan 1996 17:29:26 -0800 (PST)
To: forrerj@ucs.orst.edu
Cc: hfsig@tapr.org
Message-ID: <199601310129.RAA17383@servo.qualcomm.com>
Rendered body type: text/plain

>So I gather all that happens is that the coding gain takes a hit as
>one starts puncturing, but assuming one use it only when there are
>adequate S/N, that should be OK. Besides the little bit of additional
>overhead, sounds like a very clever idea to increase throughput.

Right. As you add the previously punctured symbols, coding gain
increases (because the effective rate of the code decreases) *and*
Eb/N0 increases because you're investing more total transmitter energy
in each data bit. So you win in two separate ways. This makes Type II
hybrid ARQ/FEC protocols *much* stronger than simple retransmission
because in the former you keep accumulating bit energy; you don't
discard all the energy in the first transmission just because it isn't
perfect.

Phil

56. Re: [HFSIG:832] Re: Chirp Waveforms

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 30 Jan 1996 17:59:55 -0800 (PST)
To: "Bergzen Hakan, KARL" <Hakan.Bergzen@telub.se>
Cc: hfsig@tapr.org
Message-ID: <199601310159.RAA17462@servo.qualcomm.com>
Rendered body type: text/plain

>It is the simplicity of coping with narrowband interfernce by inserting
>notch filters
>that is one of its strengths. You split the the incoming signal to pass x

This is in principle true for any well-designed modulation system that
uses excess bandwidth. True spread spectrum systems clearly qualify,
but so do heavily coded narrowband systems. In many cases you can just
insert a "pre-whitening" filter ahead of the demodulator that will take
out narrowband CW interference.

Phil

57. PSR Report

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Wed, 31 Jan 1996 12:46:12 -0800
To: hfsig@tapr.org
Message-ID: <13303381D70@frl.orst.edu>
Rendered body type: text/plain

Hi all,

It is that time again to tell the rest of the amateur community what we
have been doing. I attach a copy of my preliminary write-up for PSR.

I would greatly appreciate any feedback, corrections, or items for
inclusion.

Thanks for your help.

--Johan

HF Special Interest Group Activities (HFSIG)
January 1996

HFSIG experienced a relatively quiet period which often is a good
opportunity for experimenters to catch up a bit. I appreciate the
patience of the group during my busy time.

Several topics of interest to those involved in HF digital
communications came under discussion:

HF Channel simulator

Alexander Kurpiers, DL8AAU, has announced that he will shortly be
releasing a version of an HF channel simulator based on the
Watterson model. This version is implemented on a Texas
Instruments evaluation module that uses a TI320C26 DSP chip.

The Watterson model is a simulation of the behavior of the
ionosphere where typical propagation modes are created
artificially. The model is documented in CCIR Report 549-2 and
the test conditions are documented in CCIR Recommendation 520-1

includes conditions such as; Good, Moderate, and Poor. These CCIR documents are available from ITU-R (please see the last issue of PSR on how to access ITU-R).

This is a very exciting development - watch HFSIG for announcements and the UPLOAD area for the code. Congratulations Alexander, we are looking forward to using the code.

Slow Speed BPSK

Pawel Jalocha, SP9VRC, recently developed a new binary phase shift keyed (BPSK) modem for use on HF. This modem transmits and receives ASCII text at 30 cps and runs on the Finnish "DSP Card 4" as well as the Motorola DSP560002EVM hardware. The waveform used for this modem is designed for extremely narrow-band operation. Normally such narrow-band operation is extremely difficult due to frequency tolerances, however, the software automatically compensates for some off-frequency operation that helps a lot.

This kind of signaling have also independently been explored by some members involved in coherent CW (CCW). Cliff Butschardt, W6HDO, is congratulated for his involvement in a 900 mile distance BPSK contact in the 160 kHz experimenters band. The conditions on those frequencies are extremely challenging. I also have heard from Paul Straks, PA0OCD, who is getting ready for slow speed BPSK experiments. Paul is also actively using the DSP sound card, the DSP56002EVM software, also on CCW.

Parallel Modem Testing

Paul Straks, PA0OCD, also informed me that a number of Dutch amateurs are in the process of getting ready to test the 15-tone orthogonal frequency division (OFDM) HF modem that was developed by Pawel Jalocha, SP9VRC. This modem has a raw data rate of 2500 bps, and with error correction coding (ECC) 833 bps. The code features AX.25 compatibility and requires a minimal amount of hardware.

HF Networking Using Standard TNC's

For may organizations operating in parts of the world where there is little or no infrastructure, there obviously is a need for means to inexpensively communicate documents such as e-mail, and FAX. HF radio offers unique opportunities in this regard, however, integrating existing amateur radio TNC's to PC-based office software applications is a challenge. Several ideas on how to utilize off-the-shelf TNC's in HF networking applications was put forward such as encapsulating TCP/IP, however, this is an area where much more needs to be done.

NVIS

The use of near vertical incidence ?? (help!) (NVIS) antennas are preferred for short to medium distance local communications. This approach relies on high angle of radiation and is a natural choice for the type of networking mentioned above.

Nordic HF Conferences

Those interested in an excellent source for contemporary ideas on HF communications; browse http://www.telub.se/Radio/NordicHF. This web site contains some downloadable materials from recent NordicHF Conferences. The 1995 conference, for example, contains several papers on HF modems, DSP HF receivers, HF spread spectrum, and HF communications protocols. Thanks to Hakan Bergzen for pointing this out.

Sound Snippets for ALE, SLOWBPSK, and OFDM

I collected and composed a couple of sound files representing some of our recent work. These files are windows .WAV files and are located in the HFSIG upload area.

EVM Interface

Several have now built the EVM interface from the schematics as
published on the upload area. However, there have been requests
from a few of experimenters that need help. Please get in
touch with me at the e-mail address listed below if you are
interested in a kit of parts. I will post an updated file for
details on various part numbers and sources that will help you
get your interface together - look for further announcements on HFSIG.

Thanks again for all the interesting contributions and please consider
joining in the on-the-air testing efforts. We would love to see how
these ideas work under real conditions. 73's

Johan Forrer, KC7WW
(E-mail: forrerj@UCS.orst.edu)

February 1996

Original source file: hfsig.9602
Source: https://archive.org/details/tapr-1997
Recovered from: https://archive.org/download/TAPR-HF_1996-02/TAPR-HF_1996-02.txt

1. Re: [HFSIG:847] HFSIG digest 228

From: RedHa@aol.com
Date: Fri, 2 Feb 1996 10:32:36 -0500
To: hfsig@tapr.org
Message-ID: <960202103234_134001137@emout05.mail.aol.com>
Rendered body type: text/plain

Hi, Rick;

Tnx for both recent forwardings and the concern for our well being. The articles provide me with material for our local network association (WANA) meetings.

We were a bit more isolated than usual for about a week, as the phone line to the mobile home failed. I replaced it Wed and spent a lot of time reading the accumulated mail.

What are your thoughts on coordinating node operation: frequency, ERP, directivity, etc.?

I'm working on a plan to put wx info on the local BBS, possibly moving to a separate service located at the new LaCrosse NWS location when the ham installation there nears completion. My initial thought is to put the local Zone service from MNDOT on the BBS, with revision once or twice per day. If you haven't explored it, connect on 726 1140, user name CUHOME, and password MNDOT1. HELPOUT will get you the menu, HELPOUT (command) will get directions on using menu items, and INFO (command) will explain what each menu item provides. METRO will get you the local forecast, revised 3 or 4 times daily. Some commands are not explained well or at all. One such is ZONE. Try ZONE (2 letter state identifier). The zone info can be focused with ZONE MN/Z=95, or other state and zone identifier. It's provided by MNDOT, primarily advertised to pilots, but includes a lot of general road and wx info.

Temp here was -30 F a while ago; up to -28 now! I'd try to make ice cream, but it would probably be grainy and I'd never get the dasher out.

Snowshoes have become essential. Some of the frequently used tracks have become hard enough to negotiate in boots. I haven't had time to try the skiis. Now that the snow has firmed, they would probably be OK, but I don't think they were appropriate in 20 inches of fresh powder. Even the snowshoes sank about a foot at first.

I'm working on two proposals at once, with deadlines today and Monday for first draft. Outside chores, like keeping cars running and the road open so the XYL can escape, interfere a lot. Sometimes I consider rejecting any assignment requiring output in less than 30 days, so that I can avoid some of the conflicts.

I recommend a book, "Flyfishing Through the Midlife Crises", Howell Raines. He models his maturing philosophy of life with the anology of transitioning from the "Redneck way of fishing", to the dryfly purist who kills no fish.

Well, I better get back to work. Hope school and all are going well for you and Barb and bid you keep warm!

73 de Red

2. 8ary FSK

From: chbrain@dircon.co.uk (Charles Brain)
Date: Fri, 02 Feb 1996 21:05:09 +0000
To: hfsig@tapr.org
Message-ID: <199602022132.AA18953@felix.dircon.co.uk>
Rendered body type: text/plain

In case anyone is interested, I have included a brief T.D of my modem. It is based very heavily on Adrian Nash's Piccolo modem. It's my first DSP project so it probably not the best implementation!!!!

---------- Snip

This is a brief description of how the 8ary FSK modem has been implemented

on a TMS320C50 EVM (not DSK).

1) The input signal is sampled at 8k samples per second.
2) The signal is mixed with a sine and cos wave at 1625Hz. The 1625hz is
   generated with a lookup table, the transmit tones also use this lookup table.
   This moves the 8 tones symetrically around 0 Hz. As the signals are now
   complex their frequency sign is not lost.
3) The I and Q channel that have been generated are now low pass filtered using
   a 32 tap FIR filter with a cutoff freqeuncy of 1khz. This filter also
   decimates the signal by four and places it's output into a 16 entry
   circular buffer. The effective sample rate is now 2Ks/s.
4) The 16 samples from the I channel are placed in the 16 real input locations
   of a 16 point FFT. The 16 samples from the Q channel are placed in the 16
   complex input locations of the 16 point complex FFT.
5) The FFT is run at each fourth input sample (2Khz). The input buffer to the
   FFT is in fact 1 bit period (8ms) in length, a happy coincidance.
6) After each FFT the magnitude squared of each of the wanted frequency bins
   is calculated(square roots are expensive to calculate). Not all bins are
used.
   When a bit completely fills the input window, the bin magnitude is at a
maximum.
7) The bin with the maximum output is divided by the sum of the magnitudes of
   all the other wanted bins. This waveform contains a component at the symbol
   rate. The result is filtered with a BPF centered at 125Hz (symbol rate).
   This signal is used to lock an NCO to the symbol rate. This locking code
   may need more work as I have not yet tested it on poor channels.
8) When a symbol epoch arrives the bin with the largest magnitude is found and
   its symbol value is written to the host port.
9) Data carrier detect is generated from the bit sync 'circuit', when bit sync
   is achieved DCD is signalled to the host P.C. This provides a fast DCD which
   is essential for rapid scanning.

Notes.
Although the main part of the code operates at 8khz the FFT, magnitude
calculation,
bit sync and symbol epoch calculation occur at 2khz, i.e they are scheduled.
On the first 8k interrupt the FFT runs, next interrupt the magnitude
calculation
runs etc.
The FFT algorithm is taken from the TI manual and uses a rectangular window.
The filters were designed using PC-DSP and parts of the modem were simulated
using
PC-SIM.
At present the whole modem operates within the interrupt. Future plans are
to add
some form of whitening filter. There are plenty of spare cycles left!
It has been tested against the NTIA test waveforms (clean tones) and manages to
achieve word sync and hold it. I haven't tried the degraded tones yet.
Voting and Golay error correction are done within the P.C.

The unused frequency bins could be used for frequency/doppler correction. I am
not sure what frequency tolerance this modem requires.

Charles G4GUO

3. Re: [HFSIG:849] 8ary FSK

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 2 Feb 1996 21:22:10 -0800 (PST)
To: chbrain@dircon.co.uk (Charles Brain)
Cc: hfsig@tapr.org
Message-ID: <199602030522.VAA10891@servo.qualcomm.com>
Rendered body type: text/plain

Charles,

Your system sounds most interesting. It's pretty much along the lines
of the system I've been contemplating; it differs mainly in the
numbers. Do you have any performance figures? I'm really curious.

I have been thinking of much larger values of m, like 128 or 256, to
improve Eb/N0 performance. Coding over a noncoherent channel behaves
very differently than on a coherent channel (e.g., BPSK). You gain by
adding a little coding to both. But while the coherent channel
asymptotically decreases in Eb/N0 requirements as you decrease the
code rate, the noncoherent channel 'bottoms out' in its Eb/N0
requirements at a certain critical code rate and then increases again
as the code rate is further lowered. This is due to the inherent
nonlinearities in noncoherent demodulation; as you decrease the code
rate below the optimum you force the demodulator to work below its

threshold.

The minimum Eb/N0 obtained at the optimum code rate is a strong
function of m, the number of "tones", with larger m always giving
better performance (assuming the symbol time doesn't become so long as
to exceed the coherence time of the channel.)

Nevertheless, uncoded noncoherent m=8 is already better on the Eb/N0
score than uncoded *coherent* BPSK, which is not too shabby. I'm told
that 8-ary FSK is common in military HF modems.

I differ somewhat in my approach to synchronization. I need some sort
of synchronization vector, which I plan to interleave with the data.
Otherwise a fade at the front of the packet could clobber the sync,
leaving the rest of the packet strong but unusable.

My idea here is to slide a series of FFTs spaced at the separation of
the individual sync symbols across the incoming audio in a buffer.
The step would be perhaps 1/4 symbol, narrow enough to have a good
chance of spotting the energy in a weak packet but not so narrow as to
make the CPU requirements prohibitive.

After FFTing and taking bin energy values, I sum all the energies in
the bins corresponding to where the sync symbols should be, and from
that I subtract the energies in all the other bins. If the result is
larger than a threshold, I declare initial lock and begin a finer
search in time to maximize this value. Now I know exactly where the
packet begins and ends and also where the individual symbols begin and
end. I then run my FFTs to recover the data symbols, producing in each
case a set of soft decision values for the Viterbi or Fano decoder
that follows. This I do for each bit within a symbol by summing all
the FFT energy bins for which the bit in question is a "1" and
subtracting all those for which the bit is a "0". This gives better
soft metrics than just taking the energy bin with the largest
magnitude and spitting out the hard-decoded log2(m) data bits that
correspond to it.

For large m, I could probably improve CPU performance considerably
during sync search by just correlating with the appropriate sinusoids
in the appropriate places and subtracting off total received energy
from the sum of the correlator outputs.

With modern clocks I shouldn't have to run any kind of tracking loop
to recover symbol timing once I've acquired synchronization. In the
event of a small symbol rate mismatch between sender and receiver, the
interleaving of the sync vector with the data should help find the
best compromise timing. Of course, the size of the data block will
have to be limited to keep frequency errors from accumulating
indefinitely.

Oh, and I'll probably use a Hamming window on each symbol at both the
transmitter and receiver to control spectral shaping and to mitigate
intersymbol interference effects.

Phil

4. Re: [HFSIG:850] Re: 8ary FSK

From: chbrain@dircon.co.uk (Charles Brain)
Date: Sun, 04 Feb 1996 08:45:45 +0000
To: hfsig@tapr.org
Message-ID: <199602040913.AA26783@felix.dircon.co.uk>
Rendered body type: text/plain

>Charles,
>
>Your system sounds most interesting. It's pretty much along the lines
>of the system I've been contemplating; it differs mainly in the
>numbers. Do you have any performance figures? I'm really curious.
>
Hello Phil,
 I am afraid that at present I have no accurate way of measuring performance.
This is one of the reasons I need to do further work on the sync. The setup I am
using is a C50 EVM for the modem and a C50 DSK for a test generator. I have
plans
to use the DSK as a crude channel simulator, take the audio output of the
EVM and
set its S/N in the DSK add a random time offset (otherwise the modem doesn't
have to
search for sync) and send it back for decoding.

I am also trying to improve my simulation capabilities, this weekend I have been porting
an H.F Channel simulator written in C by Eric Johnson of NMSU to a form where I can call it from the macro language within PC-DSP, this allows me to take a signal
add multipath/fading and doppler spread and pass it on to a simulated modem.

When I get this going I will be able to investigate the effects of different window functions
on multipath.

Synchronization has proven to be the part that has given me the most trouble!

Do you plan to send a single interleaved block of data in each frame or a number of
blocks with selective repeats at the receiving end like some military modems do? Aka FS1052
Appendix B ?
Also how are you going to synchronize to the data, will it be a bi-product of the decoder
or are you going to use a synch vector such as a Barker sequence?

I assume you are not using phase information (from your description), and won't be using
Ungerbock codes. There was an interesting article, I think it was repeated in Nordic 95 about
the use of Ungerbock codes and parallel tone modems by Steve Cook. The paper originally
appeared in the IEE York conferance on H.F in 1994.

By the way I downloaded your Digital Audio Clips from your WWW page so I know what the real
KA9Q sounds like !

I look forward to hearing you future explots and maybe I'll get to port your modem to my DSP
platform one day!

Regards Charles G4GUO.

5. EXPERIMENTAL HF MODEM BEACON PROJECT

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Sun, 04 Feb 1996 16:10:00 -0800
To: HFSIG@tapr.org
Message-ID: <9602050007.AA03512@ucs.orst.edu>
Rendered body type: text/plain

Hi all,

I now have my hardware set up and ready to run different modem beacons on HF. The beacon runs in "attended" mode and by pre-arrangement. I can work on 28, 21, 14, 7 and 3.5 MHz.

The 15-tone modem has NOS capabilities, so a connection is possible. The SLOWBPSK modem is a beacon only, however, manual operation allows keyboard-to-keyboard QSO's. I will be happy to
activate the sytem/beacon if anyone is interested in running some tests.

-- Johan, KC7WW

6. SLOW BPSK NEWS

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Wed, 7 Feb 1996 09:54:49 -0800
To: hfsig@tapr.org
Message-ID: <1D82CB45456@frl.orst.edu>
Rendered body type: text/plain

Hi all,

Some exciting news that we want to share.

Pawel, SP9VRC is presently at CERN, Switzerland, and have been patiently listening for my (KC7WW, Monroe, Oregon) slow speed BPSK transmissions on 20 meters. My setup consists of a DSP56002EVM running Pawel's SLOWBPSK program, an an ICOM 751 on LSB mode, RF output is 10W to a Butternut vertical.

For the first time; yesterday looked like Pawel got something - not yet

confirming callsigns, but that is only a matter of time. Here is what he
e-mailed me:

Well, this time something did work. Today at 16:20 UTC I got a long lock
(I didn't need to wait long for it) and then in the garbage characters
I could clearly see CQ (with capital letters) being sent at least four
times.
They were not in a row but at a distance of some 60-80 characters.
I could see as well the space before or after them.

I was listenning with the TS-440 and I used the narrow filter
(I didn't try the wide filter at that time). I was not able to
hear any audible signal by ear.
However, I noticed there is some weak but regular CW being sent on this
or close frequency - it that you ?

I left the setup running and the output is being recorded on a file.
I will send you the file later.
I have made a schedule at 20:00 UTC on 14.180 USB voice with Danie
every day this week and weekend.

I'm sure I would capture more text if we had the FEC and good block-sync.
With asynchronous transmition one fake bit can spoil several
characters...

Pawel

Sounds like fun, does'nt it ?

--Johan, KC7WW

7. Re: [HFSIG:853] SLOW BPSK NEWS

From: Clifford Buttschardt <cbuttsch@slonet.org>
Date: Wed, 7 Feb 1996 18:25:52 -0800 (PST)
To: Johan Forrer <FORRERJ@frl.orst.edu>
Cc: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960207182038.13730B-100000@biggulp.callamer.com>
Rendered body type: text/plain

Hi Johan. Congratulations on the slow speed BPSK experiments. First, I
was not aware that you had these going on twenty meters. As you know, the
one watt BPSK beacon is running on 187.65 KHz in the experimenters band.
It was printed out 1600 miles---San Luis Obispo, CA to Aitkin, Minn.
    I mention this simply to emphasize a great deal is being done and
even I seem not able to keep up! 73 Cliff Buttschardt W6HDO

8. FW: demodulator for MSK signals

From: Jon Bloom <jbloom@arrl.org>
Date: Thu, 8 Feb 1996 11:08:42 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BAF615.CF258A40@home>
Rendered body type: text/plain

Can any of you HFSIG gurus help this fellow?

-- Jon, KE3Z
jbloom@arrl.org

From: Mr. Brooke Clarke[SMTP:brooke@pacific.net]
Sent: Wednesday, February 07, 1996 11:25 PM
To: jbloom@arrl.org
Subject: demodulator for MSK signals

I have been studying Chapter 18 of the '96 Handbook. I would like to buy
or if not available build a DSP demodulator that can demodulate the MSK
signals on the Coast Guard LF beacon stations. These stations are
transmitting Differential Global Positioning System (DGPS) data.

The lower cost MSK receivers on the market use FSK demodulators. My
application for the signals is inland where the signals are weak and I
believe I need a close to optimal demodulator. I can hear signals from a
number of beacon stations with my ear so expect that a true MSK
demodulator would give me usable data.

Could you point me toward a commercial product, a ham type kit, or
further reading material?

Thanks,
Brooke
N6GCE

9. DSP Tools

From: "Wheatley, Maurice" <mwheatle@TRSystems.com>
Date: Thu, 8 Feb 1996 11:15:13 -0500
To: "'hfsig'" <hfsig@tapr.org>
Message-ID: <c=US%a=_%p=TRSystems%l=WEBSERVER960208111516AX006D00@www>
Rendered body type: text/plain

Hi everybody,
I'm new to the SIG and just getting into this interesting DSP stuff. Was wondering if anybody has any suggestions on software design/simulation packages for dsp designs that don't require a second mortgage on the house:-)
I've got Math CAD and am looking at QEDesign.
Also how about book recommendations. Best I've found so far is Marvin Frerking's "Digital Signal Processing in Communications Systems".
Any thoughts or referals to past threads would be appreciated. Thanks!

Moe AE4JY
mwheatle@trsystems.com

10. Re: [HFSIG:856] DSP Tools

From: chbrain@dircon.co.uk (Charles Brain)
Date: Thu, 08 Feb 1996 18:07:19 +0000
To: hfsig@tapr.org
Message-ID: <199602081837.AA15851@felix.dircon.co.uk>
Rendered body type: text/plain

>Hi everybody,
>I'm new to the SIG and just getting into this interesting DSP stuff. Was wondering if anybody has any suggestions on software design/simulation packages for dsp designs that don't require a second mortgage on the house:-)
>I've got Math CAD and am looking at QEDesign.
>Also how about book recommendations. Best I've found so far is Marvin Frerking's "Digital Signal Processing in Communications Systems".
>Any thoughts or referals to past threads would be appreciated. Thanks!
>
>Moe AE4JY
>mwheatle@trsystems.com
>
>
Hello Moe,

Well I use PC-DSP and PC-SIM to design filters etc cost about $200
there www page is :-

    http://www.dspsolutions.com

I am working on modifying some HF channel simulator code written by
Eric Johnson of NMSU so I can call it from within the package.

Regards Charles G4GUO

11. Re: [HFSIG:856] DSP Tools

From: srbible@mindport.net (Steven R. Bible)
Date: Thu, 8 Feb 1996 19:15:35 -0500
To: hfsig@tapr.org
Message-ID: <199602090015.TAA27985@polaris.mindport.net>
Rendered body type: text/plain

>Hi everybody,
>I'm new to the SIG and just getting into this interesting DSP stuff. Was wondering if anybody has any suggestions on software design/simulation packages for dsp designs that don't require a second mortgage on the house:-)
>I've got Math CAD and am looking at QEDesign.
>Also how about book recommendations. Best I've found so far is Marvin Frerking's "Digital Signal Processing in Communications Systems".
>Any thoughts or referals to past threads would be appreciated. Thanks!

Hello Moe,

I really enjoy the Frerking Book. I am still studying it and I would like
to construct a DSP based receiver. Both Harris and ADI have chips that are
designed for digital IF.

Harris - http://www.semi.harris.com/

AN9509.1 Digital IF Sub Sampling Using the HI5702, HSP45116 and HSP43220

HSP50016 Digital Down Converter

TB330 Higher Speed Clock Rates Help Ease Filtering Requirements in Communication D/As

AN9401 Reducing the Minimum Decimation Rate of the HSP50016 Digital Down Converter

Analog Devices - http://www.analog.com/

ADSP-21060/ADSP-21062 Super Harvard Architecture Computers

The Frerking Book does an excellent job of bringing you up from digital basics to digital communications (not data communications which usually refers to LAN based protocols).

I spent some time surfing the web for books. I created a temporary bibliography of books on Spread Spectrum, DSP, Digital Communications, and MATLAB. You can take a look at:

http://www.mindport.net/~srbible/biblio.html

I will be moving this page to the www.tapr.org site soon. In the mean time, please take a look and give comments. You can see the books that I have found in the above catagories. You can click on the titles and you will go to the publishers web pages that describes the books in detail. I have also included some technical bookstores at the bottom of the page. I have had excellent results with the San Diego Technical Bookstore (no peculinary interest, just a happy customer).

As for DSP development platforms. Both TI (www.ti.com) and Motorola (www.motorola.com) have inexpensive DSP Starter Kits (DSK) (TI) or Evaluation Modules (EVM) (Motorola). The TI C5X DSK (p/n TMDS3200051) costs $99.00 and the Motorola EVM560002 costs $149.00. I own the later and have the former on order.

The best place to read about the EVM56002 is Johan Forrer's KC7WW August 1995 QEX article (you can order back issues from the ARRL). (BTW - Johan moderates this list. Thanks for the article Johan!)

As for matimatical simulators, there's PC-DSP for which you can download an evaluation copy from the 'net (http://www.dspsolutions.com/), and there's MATLAB. There's a book entitled "Digital Signal Processing: A Laboratory Approach Using PC-DSP, 2/e" (see my biblio web page). There's also another book for MATLAB entitled "Computer-Based Exercises for Signal Processing Using MATLAB, 1/e."

Hope this helps.

73,

- Steve, N7HPR

srbible@mindport.net
n7hpr@amsat.org
n7hpr@tapr.org

12. Re: [HFSIG:853] SLOW BPSK NEWS

From: Stephan Sykes <ssykes@ns2.emirates.net.ae>
Date: Fri, 09 Feb 1996 12:46:26 +0400
To: hfsig@tapr.org
Message-ID: <9602090849.AC26511@ns2.emirates.net.ae>
Rendered body type: text/plain

What frequency on 20 are you using?

At 11:50 AM 2/7/96 -0600, you wrote:
>Hi all,
>
>Some exciting news that we want to share.
>
>Pawel, SP9VRC is presently at CERN, Switzerland, and have been patiently
>listening for my (KC7WW, Monroe, Oregon) slow speed BPSK transmissions on
>20 meters. My setup consists of a DSP56002EVM running Pawel's SLOWBPSK

>program, an an ICOM 751 on LSB mode, RF output is 10W to a Butternut
>vertical.
>
>For the first time; yesterday looked like Pawel got something - not yet
>confirming callsigns, but that is only a matter of time. Here is what he
>e-mailed me:
>
>   ----------------------------
>Well, this time something did work. Today at 16:20 UTC I got a long lock
>(I didn't need to wait long for it) and then in the garbage characters
>I could clearly see CQ (with capital letters) being sent at least four
>times.
>They were not in a row but at a distance of some 60-80 characters.
>I could see as well the space before or after them.
>
>I was listenning with the TS-440 and I used the narrow filter
>(I didn't try the wide filter at that time). I was not able to
>hear any audible signal by ear.
>However, I noticed there is some weak but regular CW being sent on this
>or close frequency - it that you ?
>
>I left the setup running and the output is being recorded on a file.
>I will send you the file later.
>I have made a schedule at 20:00 UTC on 14.180 USB voice with Danie
>every day this week and weekend.
>
>I'm sure I would capture more text if we had the FEC and good block-sync.
>With asynchronous transmition one fake bit can spoil several
>characters...
>
>Pawel
>   ---------------------
>
>Sounds like fun, does'nt it ?
>
>--Johan, KC7WW
>
>
>
Stephan Sykes
ssykes@emirates.net.ae

13. Re: [HFSIG:859] Re: SLOW BPSK NEWS

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Fri, 9 Feb 1996 08:38:53 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960209083439.28524A-100000@ucs.orst.edu>
Rendered body type: text/plain

Hi Stephen,

We are on 14.100, closeby the international CW beacon frequency. Set
your dial for a readout of 14.100 on LSB mode. The actual carrier is some
1650 Hz lower down from 14.100. Use the CW beacons as a reference for
propation status.

The beacon is on for a few hours between about 12.00 - 20.00 UTC. I can
arrange to have it on for other periods as well.

73's

--Johan, KC7WW

On Fri, 9 Feb 1996, Stephan
Sykes wrote:

> What frequency on 20 are you using?
>
> At 11:50 AM 2/7/96 -0600, you wrote:
> >Hi all,
> >
> >Some exciting news that we want to share.
> >
> >Pawel, SP9VRC is presently at CERN, Switzerland, and have been patiently
> >listening for my (KC7WW, Monroe, Oregon) slow speed BPSK transmissions on
> >20 meters. My setup consists of a DSP56002EVM running Pawel's SLOWBPSK

> >program, an an ICOM 751 on LSB mode, RF output is 10W to a Butternut
> >vertical.
> >
> >For the first time; yesterday looked like Pawel got something - not yet
> >confirming callsigns, but that is only a matter of time. Here is what he
> >e-mailed me:
> >
> >   ----------------------------
> >Well, this time something did work. Today at 16:20 UTC I got a long lock
> >(I didn't need to wait long for it) and then in the garbage characters
> >I could clearly see CQ (with capital letters) being sent at least four
> >times.
> >They were not in a row but at a distance of some 60-80 characters.
> >I could see as well the space before or after them.
> >
> >I was listenning with the TS-440 and I used the narrow filter
> >(I didn't try the wide filter at that time). I was not able to
> >hear any audible signal by ear.
> >However, I noticed there is some weak but regular CW being sent on this
> >or close frequency - it that you ?
> >
> >I left the setup running and the output is being recorded on a file.
> >I will send you the file later.
> >I have made a schedule at 20:00 UTC on 14.180 USB voice with Danie
> >every day this week and weekend.
> >
> >I'm sure I would capture more text if we had the FEC and good block-sync.
> >With asynchronous transmition one fake bit can spoil several
> >characters...
> >
> >Pawel
> >   ---------------------
> >
> >Sounds like fun, does'nt it ?
> >
> >--Johan, KC7WW
> >
> >
> >
> Stephan Sykes
> ssykes@emirates.net.ae
>
>

14. TNX for help

From: "Wheatley, Maurice" <mwheatle@TRSystems.com>
Date: Fri, 9 Feb 1996 15:25:11 -0500
To: "'hfsig'" <hfsig@tapr.org>
Message-ID: <c=US%a=_%p=TRSystems%l=WEBSERVER960209152514AB005400@www>
Rendered body type: text/plain

Thanks to Charles and Steven for the help. The
http://www.mindport.net/~srbible/biblio.html site is a great launching point
for technical book sources. Now to figure out how to funnel all that information into my head so I can
contribute something useful someday. Thanks again.

Moe AE4JY
mwheatle@trsystems.com

15. HF channel simulator

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Sat, 10 Feb 1996 21:08:37 -0800
To: hfsig@tapr.org
Cc: kurpiers@zeus.uet.e-technik.th-darmstadt.de
Message-ID: <9602110504.AA20014@ucs.orst.edu>
Rendered body type: text/plain

Dear Alexander,

Thank you very much for your posting of the HF channel simulator. It looks
like a wonderful tool and I am sure we are only at the very beginning of
this kind of exploration, at least as amateurs.
Fortunately for me, the bit of comments in German was no problem.

I looked a bit at the code and have a few questions that I would appreciate
your comments: Which assembler was used? It obviously is not the DSK
assembler because that cannot even handle simple directives such as ".equ"
or compound statements such as "label+4". I did tried Speech Corporation's
TASM for the 320C25 as used with the DSP93, but that terminates in error

because of some internal table overflow. (If anyone could tell me how to
increase the size of the "bpoutp" I would be grateful).

Do you perhaps know of a PD assembler that would assemble the program?

The other bit of information that would be quite useful to know beforehand
is the memory requirements. It so happens that I also have an LED bargraph
at port 0 and external program memory located at 1000-EFFFh, so I may just
be lucky to be able to run the simulator on my platform without too much
problems.

Thanks again for your valued contribution.

--Johan, KC7WW

16. Re: [HFSIG:853] SLOW BPSK NEWS

From: danie.brynard@pixie.co.za (Danie Brynard)
Date: Sun, 11 Feb 1996 10:44:26 +0200
To: hfsig@tapr.org
Message-ID: <199602110846.KAA18842@f15.pix.za>
Rendered body type: text/plain

REGARDING SLOW BPSK between CERN and Pretoria (ZS):

Well all I can report is that we have not been successfull yet :-(If there
are any other guys interested in working SLOW BPSK with me let me know
please ! I can work the east coast of the USA quite easily but the west
coast seems to be a problem. (were Johan kc7ww is)

I can also hear the Italians VERY often so are their any Italian SLOW BPSK
guys out there ?

73 Danie zs6awk

>Hi all,
>
>Some exciting news that we want to share.
>
>Pawel, SP9VRC is presently at CERN, Switzerland, and have been patiently
>listening for my (KC7WW, Monroe, Oregon) slow speed BPSK transmissions on
>20 meters. My setup consists of a DSP56002EVM running Pawel's SLOWBPSK
>program, an an ICOM 751 on LSB mode, RF output is 10W to a Butternut
>vertical.
>
>For the first time; yesterday looked like Pawel got something - not yet
>confirming callsigns, but that is only a matter of time. Here is what he
>e-mailed me:
>
>   ----------------------------
>Well, this time something did work. Today at 16:20 UTC I got a long lock
>(I didn't need to wait long for it) and then in the garbage characters
>I could clearly see CQ (with capital letters) being sent at least four
>times.
>They were not in a row but at a distance of some 60-80 characters.
>I could see as well the space before or after them.
>
>I was listenning with the TS-440 and I used the narrow filter
>(I didn't try the wide filter at that time). I was not able to
>hear any audible signal by ear.
>However, I noticed there is some weak but regular CW being sent on this
>or close frequency - it that you ?
>
>I left the setup running and the output is being recorded on a file.
>I will send you the file later.
>I have made a schedule at 20:00 UTC on 14.180 USB voice with Danie
>every day this week and weekend.
>
>I'm sure I would capture more text if we had the FEC and good block-sync.
>With asynchronous transmition one fake bit can spoil several
>characters...
>
>Pawel
>   ---------------------
>
>Sounds like fun, does'nt it ?
>
>--Johan, KC7WW
>
>

>

17. Re: [HFSIG:856] DSP Tools

From: danie.brynard@pixie.co.za (Danie Brynard)
Date: Sun, 11 Feb 1996 10:44:45 +0200
To: hfsig@tapr.org
Message-ID: <199602110847.KAA18851@f15.pix.za>
Rendered body type: text/plain

The ALEFNULL stuff at nic.funet.fi is a good start ! Matlab is also a good analysis tool. There is a student version out. The next step is to get a DSP56002EVM .....73 danie zs6awk

>Hi everybody,
>I'm new to the SIG and just getting into this interesting DSP stuff. Was wondering if anybody has any suggestions on software design/simulation packages for dsp designs that don't require a second mortgage on the house:-)
>I've got Math CAD and am looking at QEDesign.
>Also how about book recommendations. Best I've found so far is Marvin Frerking's "Digital Signal Processing in Communications Systems".
>Any thoughts or referals to past threads would be appreciated. Thanks!
>
>Moe AE4JY
>mwheatle@trsystems.com
>
>

18. Re: [HFSIG:859] Re: SLOW BPSK NEWS

From: danie.brynard@pixie.co.za (Danie Brynard)
Date: Sun, 11 Feb 1996 10:45:02 +0200
To: hfsig@tapr.org
Message-ID: <199602110847.KAA18857@f15.pix.za>
Rendered body type: text/plain

>What frequency on 20 are you using?
>

I can be on 14.168 SSB/CW and 14.065 SBPSK if somebody want to try slow bpsk. A typical time suitable for me is 2000UTC. Let me know .... 73 danie zs6awk

BRYD@KIDD.CO.ZA

19. Re: [HFSIG:860] Re: SLOW BPSK NEWS

From: danie.brynard@pixie.co.za (Danie Brynard)
Date: Sun, 11 Feb 1996 10:45:05 +0200
To: hfsig@tapr.org
Message-ID: <199602110847.KAA18860@f15.pix.za>
Rendered body type: text/plain

Johan how did you determine the optimum audio level input for your SBPSK demod ?   Danie zs6awk BRYD@KIDD.CO.ZA

>Hi Stephen,
>
>
>We are on 14.100, closeby the international CW beacon frequency. Set
>your dial for a readout of 14.100 on LSB mode. The actual carrier is some
>1650 Hz lower down from 14.100. Use the CW beacons as a reference for
>propation status.
>
>The beacon is on for a few hours between about 12.00 - 20.00 UTC. I can
>arrange to have it on for other periods as well.
>
>73's
>
>--Johan, KC7WW
>

20. HF channel simulator

From: "Bergzen Hakan, KARL" <Hakan.Bergzen@telub.se>
Date: Sun, 11 Feb 96 12:55:00 GMT
To: HFSIG at TAPR <hfsig@tapr.org>
Message-ID: <3128E503@noak>
Rendered body type: text/plain

Hello,

This week I participated (only passively though) in a very good HF conference, or colloquium as they call it just to make it tricky to pronounciate and spell ;-). It was called 'HF frequency selection and management techniques for HF communications', and was held in London and arranged by the IEE.

It addresses several topics of interest to this group, e.g. the DAMSON northern latitude ionospheric measurement project, NATO work on developing a new robust waveform to handle those extreme conditions (as measured by DAMSON) and development of adaptive HF systems (and standardization thereof).

I will, however, mention two other topics of interest:

1. The next ITU frequency conference (I don't know if it really is called a conference) will discuss a Swedish proposal of using frequency block allocations on MF and HF wavelengths thus enabling truly efficient use of the new adaptive HF systems. (The situation today seems to be quite chaotic) A paper by L W Barcaly called 'A changed regulatory environment for the HF services' addressed this. I guess you could say that the amateur radio allocations are the truly first block allocations.

2. Tricia Willink of the Communications Research Centre in Ottawa presented their work on validating HF channel simulators being developed by both them and the UK Defence Research Agency. The paper is called 'Validation of HF channel simulators'. These were very thorough and professional tests and I would think it wise to consider a likewise approach if an HF channel simulator would be developed within this group (e.g. based on Alexander's simulator). Thus we could truly say that it behaves as expected.

A brief report from

Hakan (hakan.bergzen@telub.se)

21. Re: [HFSIG:858] Re: DSP Tools

From: Phil Karn <karn@qualcomm.com>
Date: Sun, 11 Feb 1996 18:44:50 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199602120244.SAA12131@qualcomm.com>
Rendered body type: text/plain

At 06:22 PM 2/8/96 -0600, you wrote:

>As for DSP development platforms. Both TI (www.ti.com) and Motorola
>(www.motorola.com) have inexpensive DSP Starter Kits (DSK) (TI) or
>Evaluation Modules (EVM) (Motorola). The TI C5X DSK (p/n TMDS3200051) costs
>$99.00 and the Motorola EVM560002 costs $149.00. I own the later and have
>the former on order.

Don't forget that you can do serious DSP on general purpose computers without a DSP board. Pentiums and the faster 486s rival many dedicated DSPs, especially on floating point computations.

Phil

22. Re: [HFSIG:868] Re: DSP Tools

From: danie.brynard@pixie.co.za (Danie Brynard)
Date: Mon, 12 Feb 1996 07:06:56 +0200
To: hfsig@tapr.org
Message-ID: <199602120509.HAA14738@f15.pix.za>
Rendered body type: text/plain

>Don't forget that you can do serious DSP on general purpose computers without
>a DSP board. Pentiums and the faster 486s rival many dedicated DSPs, especially
>on floating point computations.
>

Sounds exciting Phil. What software is available so far for the satellites ?

73 Danie

23. Re: [HFSIG:853] SLOW BPSK NEWS

From: "Helmut Strickner" <hstr@gil.com.au>
Date: Mon, 12 Feb 1996 23:35:46 +1000
To: hfsig@tapr.org
Message-ID: <199602121335.IAA29492@iccu6.ipswich.gil.com.au>
Rendered body type: text/plain

>
Hello Johan, Pawel and all

good news about the slow BPSK. I would be interested in participating
in the experiments. Unfortunately the 10 hour time difference between
VK land and stateside/Europe is a bit of a problem with listening to
the beacon (also the propagation). Johan I wonder if you could have the
beacon on weekends when there is propagation to VK4?
The other question is regarding the modem software, I have not seen
it as yet, probably Pawel distributed it only to a small group.
Perhaps I could get a copy too, I would be able to put out a beacon
from my QTH during some periods of propagation if there is any
interest.

Helmut VK4STR

 Hi all,
>
> Some exciting news that we want to share.
>
> Pawel, SP9VRC is presently at CERN, Switzerland, and have been patiently
> listening for my (KC7WW, Monroe, Oregon) slow speed BPSK transmissions on
> 20 meters. My setup consists of a DSP56002EVM running Pawel's SLOWBPSK
> program, an an ICOM 751 on LSB mode, RF output is 10W to a Butternut
> vertical.
>
> For the first time; yesterday looked like Pawel got something - not yet
> confirming callsigns, but that is only a matter of time. Here is what he
> e-mailed me:
>
>   ----------------------------
> Well, this time something did work. Today at 16:20 UTC I got a long lock
> (I didn't need to wait long for it) and then in the garbage characters
> I could clearly see CQ (with capital letters) being sent at least four
> times.
> They were not in a row but at a distance of some 60-80 characters.
> I could see as well the space before or after them.
>
> I was listenning with the TS-440 and I used the narrow filter
> (I didn't try the wide filter at that time). I was not able to
> hear any audible signal by ear.
> However, I noticed there is some weak but regular CW being sent on this
> or close frequency - it that you ?
>
> I left the setup running and the output is being recorded on a file.
> I will send you the file later.
> I have made a schedule at 20:00 UTC on 14.180 USB voice with Danie
> every day this week and weekend.
>
> I'm sure I would capture more text if we had the FEC and good block-sync.
> With asynchronous transmition one fake bit can spoil several
> characters...
>
> Pawel
>   ---------------------
>
> Sounds like fun, does'nt it ?
>
> --Johan, KC7WW
>
>
>
>

24. Re: [HFSIG:866] Re: SLOW BPSK NEWS

From: dona@tridelta.com (Don Adams)
Date: Mon, 12 Feb 1996 09:14:56 -0500 (EST)
To: hfsig@tapr.org
Message-ID: <199602121414.JAA24420@sparcy.tridelta.com>
Rendered body type: text/plain

Where's the spec's on the format Ur using for this slow bsk?
(I've read most of the archives but I must have missed it)

                over and out don   WA8QZZ

25. Re: [HFSIG:870] Re: SLOW BPSK NEWS

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 12 Feb 1996 08:59:04 -0800
To: hfsig@tapr.org
Message-ID: <24F41D42498@frl.orst.edu>
Rendered body type: text/plain

Hello Helmuth,

I would be happy to put the beacon on for longer/different periods and that would be nice to get your participating in listening. Which times would you suggest trying?

This weekend I heard the JA CW beacon in the late afternoons, so I suppose there is propagation towards the east.

When I get home tonight I'll forward you my copy of SLOWBPSK. I am in the process of putting together an updated collection of items for the EVM for use on HF - that will probably be ready in the near future. This includes some new stuff for the 66/80 MHz EVM. I will ask Pawel whether it is OK for me to include his latest code. These are the PSK terrestrial/satellite modems and the SLOWBPSK. I also have had excellent results using his FAX/SSTV DSP modem.

Till later,

--Johan, KC7WW

26. Re: [HFSIG:871] Re: SLOW BPSK NEWS

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 12 Feb 1996 09:07:57 -0800
To: hfsig@tapr.org
Message-ID: <24F67D9386A@frl.orst.edu>
Rendered body type: text/plain

Don,

The SLOWBPSK format was described in several of Pawel's (SP9VRC) postings. There was some discussion about the use and implementation of "raised cosine" pulse shaping - if you could locate that, you will see what it's about.

But just to elaborate a bit if I may, it is BPSK at 30 bps. The symbol set is straight ASCII. Nothing fancy as far as this is concerned. Wafeform shaping and a good demodulator makes the difference.

Hope this helps. Perhaps Pawel want to elaborate further. How about that Pawel?

--Johan

27. DSP processors

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Mon, 12 Feb 1996 09:17:50 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199602121717.AA000935471@polecat.sr.hp.com>
Rendered body type: text/plain

Phil Karn <karn@qualcomm.com> wrote:

>Don't forget that you can do serious DSP on general purpose computers without
>a DSP board. Pentiums and the faster 486s rival many dedicated DSPs, especially
>on floating point computations.

And for non-real-time processing (or for simple low-bandwidth applications) you can make do with something a lot less powerful than a Pentium.

Al N1AL

28. Help with EZKIT

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 12 Feb 1996 09:25:55 -0800
To: HFSIG@tapr.org
Message-ID: <24FB4927CC6@frl.orst.edu>
Rendered body type: text/plain

Hi,

I wonder if there is some kind soul that could help me get my EZKITLITE going - My second software disk is bad. As far as I can tell, all I need

is the file "EZKITAPP.EXE" and it is called "EZKITAPP.EX$" on the disk.

I would greatly appreciate it if someone can e-mail me the file. It is some
170K, otherwise I have to go the long route to get it replaced.

Thanks much.

--Johan

29. Re: [HFSIG:872] Re: SLOW BPSK NEWS

From: Clifford Buttschardt <cbuttsch@slonet.org>
Date: Mon, 12 Feb 1996 10:35:56 -0800 (PST)
To: Johan Forrer <FORRERJ@frl.orst.edu>
Cc: hfsig@tapr.org, ss@tapr.org
Message-ID: <Pine.OSF.3.91.960212102003.1931D-100000@biggulp.callamer.com>
Rendered body type: text/plain

Hi Johan. Many thanks for your encouragement regarding the slow speed
BPSK that we both are doing. I was most interested to see where your are
using 30 bps ASCII! Since I had no clue, it is refreshing to know this.
We are using ten bps ASCII also with seven data bits and two stop bits.
This produces one character persecond. The software is entirely done,
and in fact improved daily. We started out with 12 samples per bit and
now are up to 20----the maximum that a 486 computer can analyze real
time. The system is about to be changed to what is called PSKL which
stands for Phase Shift Keying Lattice code. Where the 7 bit code and a
two to the seventh or 128 possible pattern, PSKL proposes to transmit a 16
bit frame for each character instead of ten. Here is where things differ
from anything normal. 128 characters have to be selected. Finally Bill
DeCarle came up with 128 codewords that differ from each other by at
least six bit positions!! After all this is done then the computer
selects the one character that has the smallest number of altered bits.
     Now before I scare everyone off with all this, realize that the work
done so far is ORDINARY ASCII----just as you are using. The PSKL is
simply something we are doing for future evaluation.
     Maybe now it becomes more apparent why I have no patience fooling
around with a silly bulletin board system that will not work for me!!
Cliff Buttschardt   W6HDO

30. Re: [HFSIG:869] Re: DSP Tools

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 12 Feb 1996 15:41:57 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199602122341.PAA17640@servo.qualcomm.com>
Rendered body type: text/plain

>Sounds exciting Phil. What software is available so far for the satellites ?

I haven't written anything yet. I'm working on a m-ary FSK scheme that's
designed primarily for HF. Because it has pretty stringent frequency
stability requirements, I doubt it will work well on satellite channels.

Phil

31. Thanks for help on EZKITAPP

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 12 Feb 1996 16:13:32 -0800
To: HFSIG@tapr.org
Message-ID: <25680386673@frl.orst.edu>
Rendered body type: text/plain

Hi,

Thanks for all the help with obtaining a good copy of the EZKITLITE
file. Marv was kind enough to help me out.

73's

--Johan, KC7WW

32. RE: [HFSIG:853] SLOW BPSK NEWS

From: "Sadowski,P O4 USAF JCSAR/EDD" <sadowskp@jcsar.nellis.af.mil>
Date: Wed, 07 Feb 96 11:12:00 cst
To: hfsig <hfsig@tapr.org>
Message-ID: <311FCF59@mailgtwy.nellis.af.mil>
Rendered body type: text/plain

CONGRATS Pawel and Johan... keep up the good work... especially the
FEC to make it more error free....

BTW... I understand the DSP eval board you are using has been upgraded
by Motorola to a faster version 50MHz to 80MHz... any impact???

ALOHA
AL AH6LS

From: hfsig
To: hfsig
Subject: [HFSIG:853] SLOW BPSK NEWS
Date: Wednesday, February 07, 1996 11:50AM

Hi all,

Some exciting news that we want to share.

Pawel, SP9VRC is presently at CERN, Switzerland, and have been patiently
listening for my (KC7WW, Monroe, Oregon) slow speed BPSK transmissions on
20 meters. My setup consists of a DSP56002EVM running Pawel's SLOWBPSK
program, an an ICOM 751 on LSB mode, RF output is 10W to a Butternut
vertical.

For the first time; yesterday looked like Pawel got something - not yet
confirming callsigns, but that is only a matter of time. Here is what he
e-mailed me:

Well, this time something did work. Today at 16:20 UTC I got a long lock
(I didn't need to wait long for it) and then in the garbage characters
I could clearly see CQ (with capital letters) being sent at least four
times.
They were not in a row but at a distance of some 60-80 characters.
I could see as well the space before or after them.

I was listenning with the TS-440 and I used the narrow filter
(I didn't try the wide filter at that time). I was not able to
hear any audible signal by ear.
However, I noticed there is some weak but regular CW being sent on this
or close frequency - it that you ?

I left the setup running and the output is being recorded on a file.
I will send you the file later.
I have made a schedule at 20:00 UTC on 14.180 USB voice with Danie
every day this week and weekend.

I'm sure I would capture more text if we had the FEC and good block-sync.
With asynchronous transmition one fake bit can spoil several
characters...

Pawel

Sounds like fun, does'nt it ?

 --Johan, KC7WW

33. Re: [HFSIG:879] RE: SLOW BPSK NEWS

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Mon, 12 Feb 1996 23:44:58 -0800
To: hfsig@tapr.org
Message-ID: <9602130740.AA29098@ucs.orst.edu>
Rendered body type: text/plain

Hi Al,

>
>CONGRATS Pawel and Johan... keep up the good work... especially the
>FEC to make it more error free....

Well, we are not sure that we actually made contact. We will know for sure
one day when we see the callsigns in print. Anyway we are still trying.

Yes - FEC could make the difference. A good sync vector is probably needed
as well, and with the low signals, probably a lower the symbol rate as well.
But its a good start.

>
>BTW... I understand the DSP eval board you are using has been upgraded
>by Motorola to a faster version 50MHz to 80MHz... any impact???

I have gotten one of the new 66 MHz units - actually it is now the standard.
It appears to run at 80 MHz, at least the programs I tried seem to run with
no apparent problems. This is the unit that is running the beacon test -
appears to be quite stable.

--Johan

34. SLOWBPSK data format

From: Pawel Jalocha <JALOCHA@vxcern.cern.ch>
Date: Tue, 13 Feb 1996 11:32:43 +0100 (CET)
To: hfsig@tapr.org
Message-ID: <960213113243.21b2422c@vxcern.cern.ch>
Rendered body type: text/plain

Hello to all,

The SLOWBPSK.ASM makes an audio signal at 1650 Hz (defineable) which is
DBPSK modulated at 31.25 bits/sec. The 31.25 comes from the 8000 Hz
sampling which I divide by 256.

Bit format: phase transition => logical 1
            no phase transition => logical 0

Data encoding: asynchronous-like 7-bit ASCII:
start-bit + seven data bits + stop bit.
But note that the bit stream is still synchronous !

The receiver when not locked slowly sweeps a defineable audio range
(like 1600-1700 Hz at 3 Hz/sec) and when locked it stops sweeping
and decodes the data while still tracking the carrier.

Obviously I am thinking of adding a FEC... my major problem
is how to organize the blocks and how to synchronize to these
at the receiver.

Pawel

35. Re: 56002 EVM

From: chbrain@dircon.co.uk (Charles Brain)
Date: Tue, 13 Feb 1996 18:22:59 +0000
To: hfsig@tapr.org
Message-ID: <199602131852.AA19557@felix.dircon.co.uk>
Rendered body type: text/plain

Hello Johan,
>I have gotten one of the new 66 MHz units - actually it is now the standard.
>It appears to run at 80 MHz, at least the programs I tried seem to run with
>no apparent problems. This is the unit that is running the beacon test -
>appears to be quite stable.
>
>--Johan
>
>
Do you have the part number for the EVM board. Is the 56002 a superset
of the 56000 ?

73 Charles G4GUO

36. Re: [HFSIG:882] Re: 56002 EVM

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Tue, 13 Feb 1996 13:04:04 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960213125531.6810A-100000@ucs.orst.edu>
Rendered body type: text/plain

Hi Charles,

I don't want to sound like a Moto salesman - just my personal experiences
for what it's worth.

On Tue, 13 Feb 1996, Charles Brain wrote:

> Hello Johan,
> >I have gotten one of the new 66 MHz units - actually it is now the standard.

> >It appears to run at 80 MHz, at least the programs I tried seem to run with
> >no apparent problems. This is the unit that is running the beacon test -
> >appears to be quite stable.
> >
> >--Johan
> >
> >
> Do you have the part number for the EVM board. Is the 56002 a superset
> of the 56000 ?
>
> 73 Charles G4GUO
>
>

The part is still called the DSP56002EVM, but be sure to ask to make
sure that the DSP has the "66" stamped on it. That is how you would know.

The 56002 is part of the 56000 family and there are many similarities
shared by products in the family. The 56002 uses a different die
than the 56001, the new process giving it the higher clocking capability.
I would not call it a "superset" instruction set, but rather a few extra
bits in some of the special registers that used to be labelled "reserved".

Does this help?

--Johan

37. Re: [HFSIG:882] Re: 56002 EVM

From: Marijan Miletic <s56a@s55tcp.ampr.org>
Date: Tue, 13 Feb 96 21:38:15 EST
To: hfsig@tapr.org
Message-ID: <139129@s55tcp.ampr.org>
Rendered body type: text/plain

[Empty message body]

38. Re: [HFSIG:883] Re: 56002 EVM

From: chbrain@dircon.co.uk (Charles Brain)
Date: Tue, 13 Feb 1996 21:22:18 +0000
To: hfsig@tapr.org
Message-ID: <199602132151.AA00748@felix.dircon.co.uk>
Rendered body type: text/plain

>
>The part is still called the DSP56002EVM, but be sure to ask to make
>sure that the DSP has the "66" stamped on it. That is how you would know.
>   ^^
Hi Johan,
Yes that helps, Its difficult to explain to people at the other end of the
phone the differences in these things.

I guess I'll get one of these as it seems to be the platform of choice for
HFSIG, better than the TI DSK. I only asked about the superset !? as I
have a 56000 assembler manual upstairs so I can look at that.

Tnx again

Charles G4GUO

39. New EVM library available

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Wed, 14 Feb 1996 09:33:23 -0800
To: hfsig@tapr.org
Message-ID: <27FD5DB54C8@frl.orst.edu>
Rendered body type: text/plain

Hi,

I have updated the DSP56002EVM software library with the more recent
versions of code. It is now available for downloading as
EVM56K2A.TXT and EVM56K2A.ZIP from the tapr/SIG/hfsig/upload area.

As usual, this is the full source code to everything that was contributed.

Thank you to those that generously contributed their fine efforts - we
are most appreciative.

Enjoy.

--Johan, KC7WW

40. Sync Vectors for HF Digital

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Wed, 14 Feb 1996 14:06:58 -0800
To: HFSIG@tapr.org
Message-ID: <28465840951@frl.orst.edu>
Rendered body type: text/plain

Hi all,

I am looking for ideas on a suitable sync. vector for use on HF,
something better than what is currently used on Packet or Pactor for
instance.

In this regard, AX.25 Packet use "7Eh", Pactor uses a "55h/AAh"
pattern, the so-called "header byte." BTW not really for frame sync
purposes - that is done by brute force search for a valid CRC over a
block of raw bits.

Phil suggests a 63-bit PN sequence in his "Toward New Link Layer Protocols"
where a 13/64 match is considered a hit. That is some 8 bytes long - is
this the kind of header that should be used.

Any suggestions or ideas would be welcome.

Thanks in advance.

--Johan, KC7WW

41. Re: [HFSIG:887] Sync Vectors for HF Digital

From: chbrain@dircon.co.uk (Charles Brain)
Date: Thu, 15 Feb 1996 07:22:27 +0000
To: hfsig@tapr.org
Message-ID: <199602151012.AA01304@felix.dircon.co.uk>
Rendered body type: text/plain

>Hi all,
>
>I am looking for ideas on a suitable sync. vector for use on HF,
>something better than what is currently used on Packet or Pactor for
>instance.
>
>--Johan, KC7WW
>
>
Hello Johan,
You could use Barker codes, the HF modem we have at work use them for
sync. These codes are described on page 10-61 of "The ARRL Spread
Spectrum Source Book". In fact appendix B of that book has a whole
section on synch after clock recovery.

73 Charles

42. Re: [HFSIG:887] Sync Vectors for HF Digital

From: "Jim Lynch" <k4gvo@america.net>
Date: Thu, 15 Feb 1996 05:40:49 +0000
To: hfsig@tapr.org
Message-ID: <199602151042.EAA06195@tapr.org>
Rendered body type: text/plain

> Date: Wed, 14 Feb 1996 16:17:59 -0600 (CST)
> Reply-to: hfsig@tapr.org
> From: "Johan Forrer" <FORRERJ@frl.orst.edu>
> To: hfsig@tapr.org
> Subject: [HFSIG:887] Sync Vectors for HF Digital

> Hi all,
>
> I am looking for ideas on a suitable sync. vector for use on HF,
> something better than what is currently used on Packet or Pactor for
> instance.
>
> In this regard, AX.25 Packet use "7Eh", Pactor uses a "55h/AAh"
> pattern, the so-called "header byte." BTW not really for frame sync
> purposes - that is done by brute force search for a valid CRC over a
> block of raw bits.

>
> Phil suggests a 63-bit PN sequence in his "Toward New Link Layer Protocols"
> where a 13/64 match is considered a hit. That is some 8 bytes long - is
> this the kind of header that should be used.
>
> Any suggestions or ideas would be welcome.
>
> Thanks in advance.
>
> --Johan, KC7WW
Hi, Johan.

Would it make sense to have the header adaptive? I like Phil's idea,
but it might be a waste of time for very good connections. How about
a one byte header for great connections, no retries, a 4 byte header
for fair connections and an 8 byte for the I-don't-hear-anyone
connections.

Jim.
>
>
Jim Lynch   Fayetteville, GA   K4GVO
'86 GL1200I   k4gvo@america.net   44.36.34.20

43. Re: [HFSIG:889] Re: Sync Vectors for HF Digital

From: dona@tridelta.com (Don Adams)
Date: Thu, 15 Feb 1996 08:35:28 -0500 (EST)
To: hfsig@tapr.org
Message-ID: <199602151335.IAA28704@sparcy.tridelta.com>
Rendered body type: text/plain

>
>Would it make sense to have the header adaptive? I like Phil's idea,
>but it might be a waste of time for very good connections. How about
>a one byte header for great connections, no retries, a 4 byte header
>for fair connections and an 8 byte for the I-don't-hear-anyone
>connections.
>
>Jim.
>>
>>
>Jim Lynch   Fayetteville, GA   K4GVO
>'86 GL1200I   k4gvo@america.net   44.36.34.20
>
>
In the old IBM bisync world, the sync was esablished as the first non sync
character
afer a string >1 sync characacters.   This idea is simple and would allow U to
integrate (corrolate) over many (8 or 4 or what ever) when establishing the
contact
and then shorten it as need be for appropiate signal quality. The sync
char, itself,
(I think it was a control V or there abouts), had good corrolation qualities
of its own.

44. Re: [HFSIG:888] Re: Sync Vectors for HF Digital

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Thu, 15 Feb 1996 09:06:46 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960215085929.31390C-100000@ucs.orst.edu>
Rendered body type: text/plain

Charles,

On Thu, 15 Feb 1996, Charles Brain wrote:

> Hello Johan,
> You could use Barker codes, the HF modem we have at work use them for
> sync. These codes are described on page 10-61 of "The ARRL Spread
> Spectrum Source Book". In fact appendix B of that book has a whole
> section on synch after clock recovery.
>
> 73 Charles
>
>

I have been meaning to buy that book - this weekend at the swap I'll get
it. Thanks for the pointer.

Last night I did a bit of reading up and see the Barker sequence has very
good properties. The main issue is the autocorrelation function that give
a high only at the perfect sync position and next to nil when offset by
any other amount. The book also stated that the right PN sequence may be
as good as a Barker sequence in many situations. The maximum length
known Barker sequence is only 13 bits long.

Thanks to Kok Chen that also pointed this out.

--Johan

45. Re: [HFSIG:891] Re: Sync Vectors for HF Digital

From: Frode Weierud <frode@dxcern.cern.ch>
Date: Fri, 16 Feb 1996 12:32:06 +0100 (MET)
To: hfsig@tapr.org
Message-ID: <Pine.ULT.3.91.960216121418.29122A-100000@dxcern.cern.ch>
Rendered body type: text/plain

On Thu, 15 Feb 1996, Johan Forrer wrote:

> Last night I did a bit of reading up and see the Barker sequence has very
> good properties. The main issue is the autocorrelation function that give
> a high only at the perfect sync position and next to nil when offset by
> any other amount. The book also stated that the right PN sequence may be
> as good as a Barker sequence in many situations. The maximum length
> known Barker sequence is only 13 bits long.
>
> Thanks to Kok Chen that also pointed this out.
>

I have promised to give Johan some references on synchronisation that I
have recently been looking at. As I suppose others might be interested as
well, I will post them here.

There are specially three references dealing with Code-assisted bit
synchronisation (CABS) that combine synchronisation and error-correction
coding.

P. Couton, C. Hannaford and B. Honary, "Coding for both protection and
synchronisation", IEE Proc. Commun. Vol. 142, No.6, Dec. 1995, pp.352-356.

This is a more general paper on the use of CABS with other waveforms than
MFSK, which is the subject of the next two papers.

B. Honary, F. Zolghadr and M. Darnell, "Code-assisted bit synchronisation
(CABS) scheme", Electronic Letters, vol.25, No.2, pp.152-153.

B. Honary, F. Zolghadr, M. Darnell and M. Maundrell, "Improved HF modem
using convolutional coding", Electronics & Communication Engineering
Journal, Vol.4, No.2, April 1992, pp.73-82.

Concerning Barker sequences and word synchronisation generally the next
paper has a very good analysis.

M. Grayson and M. Darnell, "Extended analysis of word synchronisation
using sequence correlation techniques", IEE Proc. Commun. Vol. 142, No.6,
Dec. 1995, pp.357-362.

Johan was earlier asking for references on punctured codes. I know I have
some other references somewhere, but here is at least one that I have
briefly had a look at last year.

Jack Keil Wolf and Ephraim Zehavi, "P^2 Codes: Pragmatic Trellis Codes
Utilizing Punctured Convolutional Codes", IEEE Communication Magazine,
Feb. 1995, pp.94-99.

Have an enjoyable read, :-)

73's Frode

Frode Weierud
CERN, SL
CH-1211 Geneva 23, Switzerland
Phone : 41 22 7674794
Fax : 41 22 7679185
E-mail : frode@dxcern.cern.ch

46. Re: [HFSIG:892] Re: Sync Vectors for HF Digital

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Fri, 16 Feb 1996 09:53:56 -0800

To: hfsig@tapr.org
Message-ID: <9602161748.AA11073@ucs.orst.edu>
Rendered body type: text/plain

Dear Frode,

Thank you for the references. I have seen some of those and they are quite interesting.

>I have promised to give Johan some references on synchronisation that I
>have recently been looking at. As I suppose others might be interested as
>well, I will post them here.
>
>There are specially three references dealing with Code-assisted bit
>synchronisation (CABS) that combine synchronisation and error-correction
>coding.
>

As you might have guessed - I am in the process of putting together some rudimentary definition for use as a data link layer. I like Phil's general proposal in this regard with a few provisions to make it work on HF. I then propose to define it as:

1) Hybrid type II ARQ. - frame/cycle time timing fixed.
   More on the Hybrid ARQ later.
2) Convolutional code with constraint length 7.
3) Fixed symbol rate - data rate variable by use of punctured
   code in addition to data compression.

It appears that there are some choices for the frame sync sequence. The 13-bit Barker sequence looks like a good choice - unless someone has a good argument against it. Detecting the sequence or deriving bit timing, is where CABS may be used with additional advantage as it is Viterbi decoder. Experience has taught us, however, that bit sync for HF data must be a dynamic process that is applied throughout the bit stream and one cannot rely only on timing achieved at the sync vector. The sole purpose of the sync vector is to find the beginning of the frame. This is particularly valid when dealing with frames of any significant time duration.

The next question is then:

What should one use for the reverse channel signaling, i.e., the ACK's and NAK's etc.
I understand that longish (12 bits for example) PN sequences works reasonably well.

For those that are not familiar with what these are used for - when the sending station sends a data frame, the receiver need to signal back that it decoded the previous frame correctly and awaits the next frame. Or if it decoded an error, in the type II scheme, the sender will send some more coding bits, not the whole frame - this will help the receiver to "patch up the frame" using error-correction techniques (the incentive being that much shortened frames are sent, and thus increased throughput results). A problem exsists, however, when the sending station fails to read the ACK/NACK (perhaps due to interference or fading, or it is garbled.) One can see that a "weak" system of ACK/NACK's , also called the "reverse channel", will tend to give low throughput or even disastrous link failure when conditions are marginal. Designing strong reverse channel signalling is as important as the forward channel coding.

OK, with that out of the way - with the similar arguments as for the sync vector selection, PN sequences have been proven to be very powerful for use in reverse channel signaling. Keep in mind that the actual ACK/NACK *CHANGES* with every frame (of course the sender and receiver uses the same PN generator algorithm so they know what to expect). The question is whether there is something better and how long a sequence do we need??

Any suggestions/critisism/ideas on the proposed scheme are welcome.

Thanks,

--Johan, KC7WW

47. Re: [HFSIG:893] Re: Sync Vectors for HF Digital

From: chbrain@dircon.co.uk (Charles Brain)
Date: Sat, 17 Feb 1996 07:22:50 +0000
To: hfsig@tapr.org
Message-ID: <199602170752.AA23485@felix.dircon.co.uk>

Rendered body type: text/plain

Hello Johan,
>a "weak" system of ACK/NACK's , also called the "reverse channel", will
>--Johan, KC7WW

Another idea to throw into the melting pot, if you send a data block and hear no ACK/NAK i.e you timeout, should you send the data block again or simply request the last ACK/NAK to be resent ? I tried this on a simple H.F protocol a while back, this protocol had a window of one and it did seem to improve things especially as the transmitted data blocks were very long, (I was using very deep interleaving) and took a long time to resend.

I guess if the data blocks are short there is no need to do this.

Regards Charles G4GUO

48. More BPSK news

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 19 Feb 1996 11:26:51 -0800
To: HFSIG@tapr.org
Message-ID: <2F9BD135F09@frl.orst.edu>
Rendered body type: text/plain

Hi all,

Some more news on BPSK experiments on HF.

Contact was made on 10.137 between Paul, PA0OCD and Timo, OH2LVZ on Sunday February 19th using the SP9VRC program. Congratulations!

Copy was about 80% with signals being weak and in the noise. I believe that signals was inaudible most of the time. Paul was using the 56002EVM and Timo the DSP4.

It is evident that the trick is how to find the correct frequency. The software will track slight off-frequency operation, however, good frequency resolution is needed on the tranceivers. Paul suggested that a good realtime FFT analyser program is good for this purpose. When used in persistance/integrate mode, these weak signals can clearly be seen.

Keep in mind that no error-correcting coding is used at the present time, its a good demodulator and the relatively long symbol times that does it. I think Pawel is ready to play with some ECC next?

I also have been hearing from Cliff, W6HDO, that he hears my beacon in the early mornings. If there is anyone here in the US interested in BPSK or high speed OFDM, please let me know so we can arrange for a sked time and frequency.

Congratulations again and keep up the good work.

--Johan, KC7WW

49. QST tidbits

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Wed, 21 Feb 1996 10:06:25 -0800
To: HFSIG@tapr.org
Message-ID: <328676C1A2D@frl.orst.edu>
Rendered body type: text/plain

Hi all,

Wonder if you had a chance to look at the latest issue of QST yet?

It contains an interesting article by Grant Zehr on using JVFAX 7.1.
The article includes the description of an analog APT convertor as well.
You might be interested to know that this same functionality and a lot more is offered by using Pawel's FSKIFACE program for the 56K EVM. For example, you also get full transmit capability for WEFAX and SSTV, if that may be of interest to you. The FSKIFACE program for the EVM is included in the latest EVM software package on the HFSIG upload area.

The other tidbit that came as a surprize. Looking closely at the advertisement of AEA's latest DSP box, I noticed that the DSP used in that product is an Analog Devices 2105 DSP. The baoard also contains a Motorola 68K family microcontroller. This arrangement appears to offer potential for all sorts of future applications.

HAL also shows their new stand-alone Clover unit. Instead of the usual PC-
plugin card, the unit is self-contained. I am not sure whether this
one uses the older 56001 design or the 320C25 design as used in the P38.
Someone may wish to comment on that.

MFJ have been offereing a DSP option for some of their HF all-
mode units, though it appears to me as if it is some kind of pre-filter DSP
arrangement instead of a DSP modem.

It certainly look like DSP is making it into a lot of amateur radio gear
these days.

73's

--Johan, KC7WW

--Johan

50. Re: [HFSIG:896] QST tidbits

From: Frode Weierud <frode@dxcern.cern.ch>
Date: Thu, 22 Feb 1996 11:09:29 +0100 (MET)
To: hfsig@tapr.org
Message-ID: <Pine.ULT.3.91.960222095839.29501A-100000@dxcern.cern.ch>
Rendered body type: text/plain

On Wed, 21 Feb 1996, Johan Forrer wrote:

> The other tidbit that came as a surprize. Looking closely at the
> advertisement of AEA's latest DSP box, I noticed that the DSP used in
> that product is an Analog Devices 2105 DSP. The baoard also contains a
> Motorola 68K family microcontroller. This arrangement appears to offer
> potential for all sorts of future applications.

To see the technical details of the DSP232 have a look at AEA's home page
URL: http://www.mvangel.com/aea/

Frode

Frode Weierud Phone : 41 22 7674794
CERN, SL Fax : 41 22 7679185
CH-1211 Geneva 23, Switzerland E-mail : frode@dxcern.cern.ch

51. (no subject)

From: (unknown sender)
Date: (date unavailable)
Rendered body type: text/plain

Date: 22 Feb 96 06:35:30 EST
From: Peter Schulze <72253.2602@compuserve.com>
To: HFSIG <hfsig@tapr.org>
Subject: Re: DSP4100
Message-ID: <960222113530_72253.2602_EHJ48-5@CompuServe.COM>

The DSP4100 is using a design similar to the 4000 boards with the 56000 as DSP.
The 68000 now is a 12Mhz version. Large FlashRoms keep all the code.
Ram size has been extended too and the long sockets are there for even larger Ram chips.
They need the power to run their new 2khz clover on the units.

Please see my complete review of the unit in the February issue of the
digital journal (http://www.iea.com/~adrs) for more details.

73
Peter TY1PS

-*

EURAF / DTS
B.P. 06-2535, Cotonou, BENIN
Tel.: +229-313779
Fax : +229-313879
72253.2602@compuserve.com
Latest DTS info on the WEB:
http://ourworld.compuserve.com/homepages/EURAF

-*

52. Re: [HFSIG:894] Re: Sync Vectors for HF Digital

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Thu, 22 Feb 1996 10:11:04 -0800
To: hfsig@tapr.org
Message-ID: <9602221805.AA27968@ucs.orst.edu>
Rendered body type: text/plain

Charles,

>Hello Johan,
>>a "weak" system of ACK/NACK's , also called the "reverse channel", will
>>--Johan, KC7WW
>
>Another idea to throw into the melting pot, if you send a data block and
>hear no ACK/NAK i.e you timeout, should you send the data block again
>or simply request the last ACK/NAK to be resent ?

This is a problem area when sending partial coded blocks. Here is how I see it: If the receiver read a block perfectly, and it's ACK reponse gets lost (times out as you say), we simply resend the last block. Although the receiving end expects the next block, it will see that the packet counter has not been incremented nor did the sync vector change and thus knows that it is a repeat. Only if the sender gets requested specifically to act upon a block received in error will it send another layer of coding bits.

Does this make sense?

>I tried this on a simple
>H.F protocol a while back, this protocol had a window of one and it did seem
>to improve things especially as the transmitted data blocks were very long,
>(I was using very deep interleaving) and took a long time to resend.
>
>I guess if the data blocks are short there is no need to do this.
>
>Regards Charles G4GUO
>
>

I do think this idea of shortened blocks, invertible codes, and punctured code for variable coding rate, should pay off in overall transmitting times. One thing though that one must guard against is a protocol that has too many default timeout counters - this is where things break down, i.e., when one gets in the middle of a speed change and for example, interference sets in during some crucial link status update stage. All it takes is one missed state and the PN sequences are out of step.

Sounds like a lot of fun.

--Johan

53. Re: [HFSIG:898] Re: DSP4100

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Thu, 22 Feb 1996 10:35:39 -0800
To: hfsig@tapr.org
Message-ID: <9602221830.AA22325@ucs.orst.edu>
Rendered body type: text/plain

Hello Peter,

Thanks for the pointer to your review article - I found it very informative. Good job!

>The DSP4100 is using a design similar to the 4000 boards with the 56000 as DSP.
>The 68000 now is a 12Mhz version. Large FlashRoms keep all the code.
>Ram size has been extended too and the long sockets are there for even
larger Ram chips.
>They need the power to run their new 2khz clover on the units.

I don't think this is a typo about the 2kHz Clover units - could you please expand a bit on what this is about. I thought HAL was pushing for 500 Hz bandwidth, but perhaps this is a new development? Four parallel Clover channels = 2 kHz bandwidth, at roughly 800 x 4 = 3200 bps. In my opinion, this is a welcome and needed step in the right direction.

Judging from your description, I can see why a better 68K controller was needed. This is one aspect that is often overlooked - the DSP front-end

hardware effort is minimal compared to what is involved in the protocol
controller.

Thanks for the contribution, Peter.

>
>Please see my complete review of the unit in the February issue of the
>digital journal (http://www.iea.com/~adrs) for more details.
>
>73
>Peter TY1PS
>
>
>-*
>EURAF / DTS
>B.P. 06-2535, Cotonou, BENIN
>Tel.: +229-313779
>Fax : +229-313879
>72253.2602@compuserve.com
>Latest DTS info on the WEB:
>http://ourworld.compuserve.com/homepages/EURAF
>-*
>
>

--Johan, KC7WW

54. Re: [HFSIG:899] Re: Sync Vectors for HF Digital

From: chbrain@dircon.co.uk (Charles Brain)
Date: Thu, 22 Feb 1996 21:02:07 +0000
To: hfsig@tapr.org
Message-ID: <199602222133.AA29500@felix.dircon.co.uk>
Rendered body type: text/plain

>it is a repeat. Only if the sender gets requested specifically to act upon a
>block received in error will it send another layer of coding bits.
>
>Does this make sense?
>
>
Yep it makes sense,

 I think some (all!) of Phil's ideas are worth while. I think he has a
postscript
 file on his www page describing them.

A lot of the complexity I have seen in STANAG and MIL-STD protocols
is to do with allowing pre-emption of traffic when higher priority stuff
comes in, something we don't have to worry about?!

I was reading a SHAPE report that came to the conclusion that adapting
the modulation scheme was far more important than changing the data link
parameters. Well that was the understanding I got from it!

Regards Charles G4GUO.

55. Re: [HFSIG:900] Re: DSP4100

From: Frode Weierud <frode@dxcern.cern.ch>
Date: Fri, 23 Feb 1996 11:24:21 +0100 (MET)
To: hfsig@tapr.org
Message-ID: <Pine.ULT.3.91.960223112226.25523A-100000@dxcern.cern.ch>
Rendered body type: text/plain

On Thu, 22 Feb 1996, Johan Forrer wrote:

> I don't think this is a typo about the 2kHz Clover units - could you please
> expand a bit on what this is about. I thought HAL was pushing for 500 Hz
> bandwidth, but perhaps this is a new development? Four parallel Clover
> channels = 2 kHz bandwidth, at roughly 800 x 4 = 3200 bps. In my opinion,
> this is a welcome and needed step in the right direction.
>

Here is some info distributed on the WUN Mailing list:

>From BrunoUSA@aol.comFri Feb 23 11:21:58 1996
Date: Wed, 24 Jan 1996 00:05:05 -0500
From: BrunoUSA@aol.com

To: wun@grove.net
Subject: [WUN] CLOVER-2000 (Q-CLOVER, Quad-CLOVER) listen-in !

If any of you are interested in "listening" to this new digital
communications mode, tune your receivers to 11.0776 MHz or 10.3976 MHz, USB
starting at 09h00 PST, Wednesday, January 24th.

We will be experimenting the "beta" CLOVER-2000 (Q-CLOVER or Quad-CLOVER as
it is also called) between our base in Escondido, California and a remote
site in Illinois.
CLOVER-2000 is similar to standard CLOVER except that it is 2KHz wide and is
comprized of 8-tones (both phase & amplitude modulated)... it sounds quite
different friom "regular" CLOVER.
We are running 100 Watts PEP into a 16 element LPDA @ 65' in California. Our
counterpart also has a LPDA... both are aimed at each other.

This will give you an opportunity of "listening" to this new "sound"... by
the way, if you monitor these frequencies, you will also hear other more
exotic modes (39-tone, ALE, serial-tone, cypher, etc.) as they are part of
our assigned HF test frequencies.

If you hear us... drop me e-mail at: bhaineau@cts.com   and I will confirm
valid reports (describe the "sounds" you are hearing... and if you do manage
to decode the traffic, you will earn extra brownie points ;-)

Good listening...

Bruno Haineault

=====
This is the WUN mailing list.
To post to the list, send a e-mail to :   wun@grove.net
To unsubscribe the list, send a mail to: majordomo@grove.net
with only "unsubscribe wun" in the BODY of the e-mail.
=====

73's Frode

Frode Weierud
CERN, SL
CH-1211 Geneva 23, Switzerland

Phone : 41 22 7674794
Fax   : 41 22 7679185
E-mail : frode@dxcern.cern.ch

56. CLOVER 2000 (Q-CLOVER)

From: k5yfw@sacdm01.kelly.af.mil (WALT DUBOSE - K5YFW)
Date: Fri, 23 Feb 96 12:54:31 -0600
To: bhaineau@cts.com
Cc: hfsig@tapr.org
Message-ID: <9602231854.AA16763@sacdm01.kelly.af.mil>
Rendered body type: text/plain

Bruno,

BRAVO

I believe you are on the _right_ track now. A couple of years ago I
corresponded with the originators of CLOVER and expressed my belief
that hams need to quickly move to multi-tone modems of 16 to 39 tones
similar to the MIL-STD-188-110c type modems. Shortly after that I
began corresponding with several members on the TAPR HFSIG.

I am particularly impressed to see that HAL is moving up to higher
bit rates and now has an external clover board...this is very
important in emergency communications.

I am presenting a program on HF digital communications to the San
Antonio Radio Club next month and will use as a major part of my
presentation the information (papers) supplied by HAL on the CLOVER
system had were re-prints of magazine articles. I would be most
interested in obtaining the same type of information about Q-CLOVER.

While I was Chief of Communications Maintenance for the 32nd
Aeromedical Evacuation Gp (and senior communicator in the AF
Aeromedical Evacuation System) I pushed for use of the
MIL-STD-188-110c modems (16/39 tone and serial tone) use, but the

price limited our obtaining the modems (about $10K at the time). Our
requirement was 2400 BPS data thruput under adverse HF conditions at
the 100 Watt power level. As deputy communications director for the
San Antonio Emergency Management Communications Gp (pronounce that
Civil Defense Communications) I see the above requirement for our use,
especially in providing command and control communications from the
Texas Gulf Coast back to San Antonio during hurricanes.

Additionally, we have a requirement to run TCP/IP on HF...3200 BPS
(assuming 1000 BPS for FEC) would meet that need. Please consider
writing encapsulation software for the CLOVER-200 (Q-CLOVER) modem
allowing it to run TCP/IP over HF.

I am looking forward to hearing from you. The top part of my tower
is still on the ground so my LP is still on the ground and I won't be
able to hear your test.

Best Regards,

Walt DuBose

==========

In Frode's message of 23 Feb 1996 at 0947 CST, he writes:
> On Thu, 22 Feb 1996, Johan Forrer wrote:
>
> > I don't think this is a typo about the 2kHz Clover units - could you please
> > expand a bit on what this is about. I thought HAL was pushing for 500 Hz
> > bandwidth, but perhaps this is a new development? Four parallel Clover
> > channels = 2 kHz bandwidth, at roughly 800 x 4 = 3200 bps. In my opinion,
> > this is a welcome and needed step in the right direction.
> >
>
> Here is some info distributed on the WUN Mailing list:
>
> >From BrunoUSA@aol.comFri Feb 23 11:21:58 1996
> Date: Wed, 24 Jan 1996 00:05:05 -0500
> >From: BrunoUSA@aol.com
> To: wun@grove.net
> Subject: [WUN] CLOVER-2000 (Q-CLOVER, Quad-CLOVER) listen-in !
>
> If any of you are interested in "listening" to this new digital
> communications mode, tune your receivers to 11.0776 MHz or 10.3976 MHz, USB
> starting at 09h00 PST, Wednesday, January 24th.
>
> We will be experimenting the "beta" CLOVER-2000 (Q-CLOVER or Quad-CLOVER as
> it is also called) between our base in Escondido, California and a remote
> site in Illinois.
> CLOVER-2000 is similar to standard CLOVER except that it is 2KHz wide and is
> comprized of 8-tones (both phase & amplitude modulated)... it sounds quite
> different friom "regular" CLOVER.
> We are running 100 Watts PEP into a 16 element LPDA @ 65' in California. Our
> counterpart also has a LPDA... both are aimed at each other.
>
> This will give you an opportunity of "listening" to this new "sound"... by
> the way, if you monitor these frequencies, you will also hear other more
> exotic modes (39-tone, ALE, serial-tone, cypher, etc.) as they are part of
> our assigned HF test frequencies.
>
> If you hear us... drop me e-mail at: bhaineau@cts.com   and I will confirm
> valid reports (describe the "sounds" you are hearing... and if you do manage
> to decode the traffic, you will earn extra brownie points ;-)
>
> Good listening...
>
> Bruno Haineault
>
>
> =====
> This is the WUN mailing list.
> To post to the list, send a e-mail to :   wun@grove.net
> To unsubscribe the list, send a mail to: majordomo@grove.net
> with only "unsubscribe wun" in the BODY of the e-mail.
> =====
>
>
> ----------
>
> 73's Frode

>
>
>   Frode Weierud   Phone   : 41
22 7674794
>   CERN, SL   Fax   : 41 22 7679185
>

57. Re: [HFSIG:893] Re: Sync Vectors for HF Digital

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 23 Feb 1996 16:45:51 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199602240045.QAA05605@servo.qualcomm.com>
Rendered body type: text/plain

>It appears that there are some choices for the frame sync sequence. The
>13-bit Barker sequence looks like a good choice - unless someone has a good
>argument against it. Detecting the sequence or deriving bit timing, is where

Well, my main objection to Barker sequences is that the longest known
Barker sequence isn't very long. Since the sync vectors are sent
without ECC, and because the ECC code rate might be very low, you need
extreme robustness in your sync vector. That argues for a fairly long
sync vector, much longer than 13 bits.

You can think of a PN sync sequence as a "spread spectrum burst" conveying
one bit of information ("the packet starts HERE") with a considerable
processing gain equal to the number of bits in the sequence. A 64-bit PN
sequence gives a processing gain of 64:1 or about 18 dB, which should be
enough to keep working reliably even at the extremely low symbol S/Ns
implied by powerful, low rate ECC.

Phil

58. EVM56002 connecting to printer port

From: "Helmut Strickner" <hstr@gil.com.au>
Date: Sat, 24 Feb 1996 10:48:59 +1000
To: hfsig@tapr.org
Message-ID: <199602240048.TAA17928@iccu6.ipswich.gil.com.au>
Rendered body type: text/plain

Hi

I have downloaded the latest evm27.zip file from the Motorola bubfile
archive (design.NET site) and found an archive in it called spec.zip.

It contains some programmes, one of them is a frequency domain plot
program which runs on the PC. In this example the FFT is performed on
the EVM56002 which is connected to the PC parallel port via a cable
from J7. There is a file in the archive called spec2.doc which contains
the description of the program and shows how to connect the Port B of
the DSP56002 which is J7 to the parallel port on the PC.

The problem is that I am unable to read this doc file, it is not in
word format but has a label at the beginning of the file <MakerFile
4.0K>. Can somebody please tell me how to read or convert this file
into a readable and printable format? It contains some graphics about
the cable connection from J7 to the parallel port.

If someone has already converted this file is it possible to upload
it to hfsig/upload directory please.

I think to ask Motorola to post this file in a 'standard' format will
take months of waiting. It took them two months to sort out the
problems with the corrupt evm27.exe file in the bubfile archive.

Helmut

59. Re: [HFSIG:904] Re: Sync Vectors for HF Digital

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Sat, 24 Feb 1996 07:26:10 +0100 (MEZ)
To: hfsig@tapr.org
Message-ID: <Pine.A32.3.91.960224072316.19236A-100000@hades.uet.e-technik.th-darmstadt.de>
Rendered body type: text/plain

On Fri, 23 Feb 1996, Phil Karn wrote:

> >It appears that there are some choices for the frame sync sequence. The
> >13-bit Barker sequence looks like a good choice - unless someone has a good

> >argument against it. Detecting the sequence or deriving bit timing, is where
>
> Well, my main objection to Barker sequences is that the longest known
> Barker sequence isn't very long. Since the sync vectors are sent
> without ECC, and because the ECC code rate might be very low, you need
> extreme robustness in your sync vector. That argues for a fairly long
> sync vector, much longer than 13 bits.
>

The longest known binary Barker sequence is 13 bits. But there are lots
of longer Generalized Barker sequences. The longest I know is 36 bits long.
As we don't have to stick to binary sequences there is a possibility to
use Generalized Barker sequences.

> You can think of a PN sync sequence as a "spread spectrum burst" conveying
> one bit of information ("the packet starts HERE") with a considerable
> processing gain equal to the number of bits in the sequence. A 64-bit PN
> sequence gives a processing gain of 64:1 or about 18 dB, which should be
> enough to keep working reliably even at the extremely low symbol S/Ns
> implied by powerful, low rate ECC.
>
> Phil
>

Ok, from a practical point of view PN sequences seem to be better.

73' Alexander>

----------------------------------+---------------------------------------
Alexander F. Kurpiers	
Institut f. Uebertragungstechnik	Voice: +49-6151-162369
u. Elektroakustik	Fax : +49-6151-165545
Merckstrasse 25	EMail: a.kurpiers@uet.th-darmstadt.de
D-64283 Darmstadt (Germany)	Hamradio: dl8aau@db0zdf.#rpl.deu.eu
----------------------------------+---------------------------------------

60. Re: [HFSIG:905] EVM56002 connecting to printer port

From: "Jim Lynch" <k4gvo@america.net>
Date: Sat, 24 Feb 1996 06:21:13 +0000
To: "Helmut Strickner" <hstr@gil.com.au>, hfsig@tapr.org
Message-ID: <199602241122.FAA21207@tapr.org>
Rendered body type: text/plain

> Date: Fri, 23 Feb 1996 19:05:35 -0600 (CST)
> Reply-to: hfsig@tapr.org
> From: "Helmut Strickner" <hstr@gil.com.au>
> To: hfsig@tapr.org
> Subject: [HFSIG:905] EVM56002 connecting to printer port

> Hi
>
> I have downloaded the latest evm27.zip file from the Motorola bubfile
> archive (design.NET site) and found an archive in it called spec.zip.
>
> It contains some programmes, one of them is a frequency domain plot
> program which runs on the PC. In this example the FFT is performed on
> the EVM56002 which is connected to the PC parallel port via a cable
> from J7. There is a file in the archive called spec2.doc which contains
> the description of the program and shows how to connect the Port B of
> the DSP56002 which is J7 to the parallel port on the PC.
>
> The problem is that I am unable to read this doc file, it is not in
> word format but has a label at the beginning of the file <MakerFile
> 4.0K>. Can somebody please tell me how to read or convert this file
> into a readable and printable format? It contains some graphics about
> the cable connection from J7 to the parallel port.
>
> If someone has already converted this file is it possible to upload
> it to hfsig/upload directory please.
>
> I think to ask Motorola to post this file in a 'standard' format will
> take months of waiting. It took them two months to sort out the
> problems with the corrupt evm27.exe file in the bubfile archive.
>
> Helmut
It sounds like that may be a Frame Maker file. I use (rarely) frame
maker at work and might be able to translate the file into another
format, but I'm not sure about the legality of re-distributing it.
I've seen a "frame viewer" somewhere recently, but don't remember

where it was. It ran on a pc, so maybe this will jog someone's
memory. I'll keep my eyes open.

Jim.
Jim Lynch   Fayetteville, GA   K4GVO
'86 GL1200I   k4gvo@america.net   44.36.34.20

61. FrameReader (Was: EVM56002 ...)

From: gc@fox.cen.com (Gary Chatters)
Date: Sat, 24 Feb 96 12:11:31 EST
To: hfsig@tapr.org
Message-ID: <9602241711.AA04594@cen.com>
Rendered body type: text/plain

>It sounds like that may be a Frame Maker file. I use (rarely) frame
>maker at work and might be able to translate the file into another
>format, but I'm not sure about the legality of re-distributing it.
>I've seen a "frame viewer" somewhere recently, but don't remember
>where it was. It ran on a pc, so maybe this will jog someone's
>memory. I'll keep my eyes open.

FrameReader is available via anonymous ftp at ftp.frame.com.
cd to /pub/techsup/product_updates/dos and get the file: reader.zip.
PKZIP is available on some other branch of the /pub/techsup tree
to unzip the file. Even in its compressed format this is a 4.7MB file.
mac and unix versions available.

Frame also has a FrameViewer which costs about $20-$30.

On a more general note: Microsoft Word, Frame, Mathematica, and Adobe
(and others) all have free or cheap viewers available.

Gary

62. Re: [HFSIG:908] FrameReader (Was: EVM56002 ...)

From: "Steven R. Bible" <srbible@mindport.net>
Date: Sat, 24 Feb 1996 18:07:16 -0500
To: hfsig@tapr.org
Message-ID: <1.5.4b11.32.19960224230716.0067414c@mindport.net>
Rendered body type: text/plain

Does anyone know of a postscript viewer/reader for Win95? And to make it
even more challenging, can print to whatever Win95 printer driver is
installed at the time. Either I have to purchase a postscript printer or
convert postscript files to adobe PDF format.

Thanks, Steve N7HPR

>FrameReader is available via anonymous ftp at ftp.frame.com.
>cd to /pub/techsup/product_updates/dos and get the file: reader.zip.
>PKZIP is available on some other branch of the /pub/techsup tree
>to unzip the file. Even in its compressed format this is a 4.7MB file.
>mac and unix versions available.
>
>Frame also has a FrameViewer which costs about $20-$30.
>
>On a more general note: Microsoft Word, Frame, Mathematica, and Adobe
>(and others) all have free or cheap viewers available.
>
>Gary
>
>
>

63. Re: Re: FrameReader (Was: EVM56002 ...)

From: "Helmut Strickner" <hstr@gil.com.au>
Date: Mon, 26 Feb 1996 09:21:43 +1000
To: hfsig@tapr.org
Message-ID: <199602252321.SAA11458@iccu6.ipswich.gil.com.au>
Rendered body type: text/plain

> Does anyone know of a postscript viewer/reader for Win95? And to make it
> even more challenging, can print to whatever Win95 printer driver is
> installed at the time. Either I have to purchase a postscript printer or
> convert postscript files to adobe PDF format.
>

> Thanks, Steve N7HPR
>
Have you tried Ghostview and Ghostscript ? They are available under
the GNU agreement on a number of sites. I am not sure whether there
is a version out yet which runs under Win95.

As for the Frame Reader programme I have spent about 4 hours
downloading it before the ftp stalled around the 4Mb mark.
The Frame Reader is freely available and the size of the file is abt.
4.6MB. Perhaps I will have another try on a luckier day.

Helmut

>
>
> >FrameReader is available via anonymous ftp at ftp.frame.com.
> >cd to /pub/techsup/product_updates/dos and get the file: reader.zip.
> >PKZIP is available on some other branch of the /pub/techsup tree
> >to unzip the file. Even in its compressed format this is a 4.7MB file.
> >mac and unix versions available.
> >
> >Frame also has a FrameViewer which costs about $20-$30.
> >
> >On a more general note: Microsoft Word, Frame, Mathematica, and Adobe
> >(and others) all have free or cheap viewers available.
> >
> >Gary
> >
> >
> >
>
>
>

64. RE: [HFSIG:910] Re: Re: FrameReader (Was: EVM56002 ...)

From: Jon Bloom <jbloom@usa.nai.net>
Date: Mon, 26 Feb 1996 05:54:01 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB040F.32048C40@jbloom.nai.net>
Rendered body type: text/plain

Helmut Strickner[SMTP:hstr@gil.com.au] wrote:
>
>
>> Does anyone know of a postscript viewer/reader for Win95? And to make it
>> even more challenging, can print to whatever Win95 printer driver is
>> installed at the time. Either I have to purchase a postscript printer or
>> convert postscript files to adobe PDF format.
>>
>> Thanks, Steve N7HPR
>>
>Have you tried Ghostview and Ghostscript ? They are available under
>the GNU agreement on a number of sites. I am not sure whether there
>is a version out yet which runs under Win95.

Yes, the 32-bit Ghostscript 3.53 runs under Win95. I use it often, along with GSView 1.4, to attach preview bitmaps to EPS files and to view Postscript files. (I don't use it to print since I have a Postscript printer, but it will print to a Windows printer.)

See http://www.cs.wisc.edu/~ghost/

--Jon

65. Re: [HFSIG:904] Re: Sync Vectors for HF Digital

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 26 Feb 1996 08:49:17 -0800
To: hfsig@tapr.org
Message-ID: <39F223B3805@frl.orst.edu>
Rendered body type: text/plain

Hi Phil,

For the same argument, I assume that reverse channel signaling for an
ARQ system needs the same considerations as that is also an unprotected
sequence?

Any suggestions on what would be suitable there?

>

> >It appears that there are some choices for the frame sync sequence. The
> >13-bit Barker sequence looks like a good choice - unless someone has a good
> >argument against it. Detecting the sequence or deriving bit timing, is where
>
> Well, my main objection to Barker sequences is that the longest known
> Barker sequence isn't very long. Since the sync vectors are sent
> without ECC, and because the ECC code rate might be very low, you need
> extreme robustness in your sync vector. That argues for a fairly long
> sync vector, much longer than 13 bits.
>
> You can think of a PN sync sequence as a "spread spectrum burst" conveying
> one bit of information ("the packet starts HERE") with a considerable
> processing gain equal to the number of bits in the sequence. A 64-bit PN
> sequence gives a processing gain of 64:1 or about 18 dB, which should be
> enough to keep working reliably even at the extremely low symbol S/Ns
> implied by powerful, low rate ECC.
>
> Phil
>
>

--Johan

66. Re: [HFSIG:896] QST tidbits

From: N0AOT@aol.com
Date: Tue, 27 Feb 1996 12:37:20 -0500
To: forrerj@frl.orst.edu
Cc: hfsig@tapr.org, amsat-bb@amsat.org
Message-ID: <960227123719_334589791@mail04.mail.aol.com>
Rendered body type: text/plain

In a message dated 96-02-21 14:08:36 EST, you write:

>Wonder if you had a chance to look at the latest issue of QST yet?
>

Hi Johan--

I haven't talked to you for ages, but have been following your comments here on the HFSIG. Lots of neat stuff going on!

Here's another tidbit from the bottom of page 183 of Jan 96 QST: Brian Beezley, K6STI, is offering an DSP RTTY package which uses the common SB-16 sound card. He describes a graphical FFT tuning indicator and many other features. He doesn't say anything about amtor/pactor, though.

Have you or any of the others tried out this software? I have a sb compatible chip in my Pentium notebook, and I realized that if it really works, all I have to do is plug my radio directly into the mike and speaker jack of the notebook, to be on the air! How's that for simple and portable?

Now if someone would just write a 9600 baud modem for the pacsats using the SB-16 and/or Pentium, I could throw out the DSP 2232 (which AEA seems to have abandoned) along with all the rest of the external modems!

73's,

--Bob Carlson, n0aot@amsat.org

67. Re: [HFSIG:913] Re: QST tidbits

From: Clifford Buttschardt <cbuttsch@biggulp.callamer.com>
Date: Tue, 27 Feb 1996 10:46:45 -0800 (PST)
To: N0AOT@aol.com
Cc: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960227104126.16938F-100000@biggulp.callamer.com>
Rendered body type: text/plain

As you, I am most interested in Brian Beesley's efforts using the sound card instead of some of the DSP units that have attempted to do 9600, Amtor/Pactor. These tasks should have already been accomplished by the DSP 93 platform by have not! If this sound card approach does work, I'd be most interested in knowing that fact AND knowing just how difficult it might be to program for BPSK, DBPSK and similar protocols.
Cliff Buttschardt   W6HDO

68. Re: [HFSIG:914] Re: QST tidbits

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Tue, 27 Feb 1996 14:11:42 -0800 (PST)

To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960227132050.11393C-100000@ucs.orst.edu>
Rendered body type: text/plain

Hi Cliff,

On Tue, 27 Feb 1996, Clifford Buttschardt wrote:

> As you, I am most interested in Brian Beesley's efforts using the sound
> card instead of some of the DSP units that have attempted to do 9600,
> Amtor/Pactor. These tasks should have already been accomplished by the
> DSP 93 platform by have not! If this sound card approach does work, I'd
> be most interested in knowing that fact AND knowing just how difficult it
> might be to program for BPSK, DBPSK and similar protocols.
> Cliff Buttschardt   W6HDO
>
>

A couple of things to consider:

I have been hearing great things about Brian's program - a nice achievement. For RTTY, however, it is OK to "copy behind" even by as much as one symbol (baud) time. That is not workable for an ARQ system where the additional delay (as much as 10 ms) will most definately upset the tight timing cycle. However, I'm sure that it is only a matter of time until when we will see a solution.

You probably are aware of Bill de Carle's, VE2IQ, work with CCW and BPSK using a PC platform for doing DSP - perhaps pushing it to the limit, however, as the processing speeds go up and prices drop, who knows, one day we might all be doing it that way. As Phil has often said, we can get a lot of amateurs interested in participating if we can do advanced DSP on a PC rather than invest in additional external equipment.

Venturing off the HF digital topic, the issue of doing a fully featured 9600 baud modems on the DSP-93 is not as obvious. I have looked into this possibility with the intention of doing this, but must agree with the DSP-93 experts that it is, to say the least, more than a challenge. And not for lack of efforts either. The DSP-93 uses a CODEC that is programmed at a fixed A/D rate that is not a multiple of the 9600 baud clock. This means that it is not possible to ever achieve perfect sync to extract exact bit edges. The option that the designers had was to oversample the signal in order to increase resolution and thus reduce the amount of bit edge jitter. The price to pay is that this eats up a lot of DSP clock tics and thus leaves little prospect for doing things like HDLC on the DSP. I hope that I have summarized this accurately - this is what Frank Perkins, one of the DSP-93 modem designers shared with me. Someone may want to comment on this.

This problem is not unique to the DSP-93 either. Have a look at the new AEA DSP-232 and you will see that they still have an HDLC chip on the board. Guess why? If you look a bit further at other 9600 baud DSP applications like for instance the DSP sound card, you see similar things such as HDLC emulation being done on the PC, however few do attempt this on the DSP.

There is an exception though. With the 9600 baud G3RUH modem for the Motorola EVM, Jarkko, OH2LNS, pulled some clever tricks out of the hat by utilizing 56K features. He has managed to implemnt both the 9600 baud modem, HDLC decoder and 19.2K KISS interface all on the DSP. In my humble opinion, this is only possible due to the 56K architecture. For instance, it has an on-chip UART that takes care of the serial interface. Anyone that has succeeded in building a bit-banger UART on a DSP where there are high rate interrupts running at the same time, knows the value of this bit of hardware. The other bit of hardware that helps here is the high speed serial interface for dealing with the CODEC that makes it possible to implement very efficient interrupt service routines. In addition, the instruction set of this chip is more like a microcontroller than a DSP, though works like a DSP. This 9600 baud modem runs on the DSP4 which has a 27 MHz clock - the latest version of the EVM has a 66 MHz chip, just to give you an idea that the chip is not strapped for clock cycles.

My $0.02, for what it is worth.

It is important to keep in mind that each platform has its advantages and disadvantages. The potential is there, none of it is easy, and the learning curve is steep. But it is inviting and ready for exploring - if you have the time and patience.

Good luck.

--Johan, KC7WW

69. Re: [HFSIG:915] Re: QST tidbits

From: fperkins@onramp.net
Date: Tue, 27 Feb 1996 20:03:37 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <199602280203.UAA22485@mailhost.onramp.net>
Rendered body type: text/plain

Hi Johan,

Here is a couple of cents more:

Re the DSP-93 and AMTOR, I have shown it receiving AMTOR using you TOR program on several of my DSP-93 videos as seen at the DCC and AMSAT conferences. The modem side was pretty easy - TOR does the hard part. I was just wondering if you had done any experimenting with the DSP-93 TOR combo.

Out of frustration in trying to define SW to do an AX.25 FCS calculation, I threw a bit of code together for the Bell 202 modem on the DSP-93, so I could push raw packets out the DSP-93 UART and get a look at the structure of the FCS. In case anyone is interested:

  All bytes in the packet EXCEPT the FCS are sent LSB 1st.

  The FCS is sent high byte first, MSB first. The AX.25 spec nicely gets around defining this by referring to an obscure CCITT spec.

Based on this quick, rough hack, I don't think there would be any problem adding HDLC and KISS to the 300 and 1200 bps modems - if someone has the time and interest to pursue it. For the reasons you explained, I don't this is practical for 9600 bps.

73 Frank WB5IPM

70. Sound Blaster

From: k6sti@n2.net (Brian Beezley)
Date: Tue, 27 Feb 1996 18:48:37 -0800
To: hfsig@tapr.org
Message-ID: <199602280248.SAA26541@ravel.n2.net>
Rendered body type: text/plain

Johan, using the Creative Labs device driver, signal delay is about 500 ms on both transmit and receive for a 6-kHz sample rate. Most of this is in the DMA buffer, the size of which cannot be made less than 4096 bytes. The Creative Labs device driver is not easy to work with. I had a lot of trouble getting it to work reliably for continuous conversion despite three books of documentation and very accessible developer support. Some crucial parts of the documentation turned out to be in error and I had to discover this by experiment. Had I known all this before I started, I certainly would have rolled my own driver. In fact, I still may.

Using the Creative Labs driver also ties you exclusively to Sound Blaster hardware. They have a very large share of the sound card market and their cards are inexpensive, so this isn't much of a problem for desktop applications. However, many notebook computers don't use the Creative Labs Vibra 16 chip. For these machines there's no way to add an alternate sound card and therefore no way to run a program written for the Creative Labs drivers. This issue has already come up for the Heard Island DXpedition that wants to take along high-performance RTTY without having to lug around external RTTY boxes.

73--Brian, K6STI
k6sti@n2.net

71. Re: QST tidbits

From: Robert.Glassey@nmp.nokia.com
Date: Wed, 28 Feb 96 11:31:30 +0000
To: hfsig@tapr.org
Message-ID: <H00002920138a6d3@MHS>
Rendered body type: text/plain

Hi Everyone,

It's about time I made myself know to more that just one or two of you, since this is a pet topic of mine. I haven't had a chance to review Brians program, so I'm not sure what it is like or what it can do, but I am also working on a RTTY/AMTOR/PACTOR soundblaster program, and am having some success, althought there is still plently of work to do.

I've been following this group for a while now and am most interested in what you guys are getting up to. As Phil has mentioned on this group a pentium/486 is quite capable of real DSP work, and in fact I have had a _very_ simple BPSK modem working on a 12 MHZ 286! Programming for the soundblaster at hardware level on a PC is not trivial, but it is quite possible, and the DMA and interupts of a PC can be used to our advantage. I program the interupt code in fixed point assembly and the UI/protocol stuff in C.

As for when my program will be released, I'm not sure, but we're talking 6 months to a year away, unless I have a lot of unexpected time on my hands. At present my program does RTTY demod only, in a very crude manner.

> As you, I am most interested in Brian Beesley's efforts using the
> sound card instead of some of the DSP units that have attempted to do

> 9600, Amtor/Pactor. These tasks should have already been
> accomplished by the DSP 93 platform by have not! If this sound card
> approach does work, I'd be most interested in knowing that fact AND
> knowing just how difficult it might be to program for BPSK, DBPSK and
> similar protocols.
> Cliff Buttschardt   W6HDO
>

Cheers,

Rob, G0VTQ

72. Re: [HFSIG:917] Sound Blaster

From: Peter Schulze <72253.2602@compuserve.com>
Date: 28 Feb 96 07:51:59 EST
To: HFSIG <hfsig@tapr.org>
Message-ID: <960228125158_72253.2602_EHJ92-2@CompuServe.COM>
Rendered body type: text/plain

Brian,

you should consider to switch to Windows, there you have a very easy API to interface with the soundcards and you dont have to worry about the make and details.
Together with the INTEL native DSP library you get a very easy no headache DSP setup.

Peter

73. Digital Journal

From: Peter Schulze <72253.2602@compuserve.com>
Date: 28 Feb 96 07:51:54 EST
To: HFSIG <hfsig@tapr.org>
Message-ID: <960228125153_72253.2602_EHJ92-1@CompuServe.COM>
Rendered body type: text/plain

>You mentioned the digital journal in a TAPR
>SIG .... who's digital journal... available in US???

The digital journal is the monthly publication of :

International Digital Radio Association
P.O. Box 2550
Goldenrod, Florida 32733-2550 -- U.S.A.
Office: +1-407-677-7000
Fax: +1-407-671-0194
Internet E-Mail Address: adrs@iea.com
WWW Page: http://www.iea.com/~adrs

The journal has its own WWW page:
http://home.earthlink.net/~n2hos/

73 Peter TY1PS

74. Re: [HFSIG:916] Re: QST tidbits

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Wed, 28 Feb 1996 08:54:31 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960228084251.27038A-100000@ucs.orst.edu>
Rendered body type: text/plain

Hi Frank,

On Tue, 27 Feb 1996 fperkins@onramp.net wrote:

> Hi Johan,
>
> Here is a couple of cents more:
>
> Re the DSP-93 and AMTOR, I have shown it receiving AMTOR using you TOR
> program on several of my DSP-93 videos as seen at the DCC and AMSAT
> conferences. The modem side was pretty easy - TOR does the hard part.
> I was just wondering if you had done any experimenting with the DSP-93
> TOR combo.

Besides just trying it out and getting it to work, I have not done much further. One of those things on the "to do list" that I hope to explore one of these days. I also need to update the Pactor.

If anyone does not know about it yet - the source code for the TOR modes that I wrote has been available in the "TORLIB.ZIP" file on the HFSIG upload area for some time. It is for the sound card DSP board, though can be used with any modem with a bit of work. Feel free to use it and share your experiences.

>
> Out of frustration in trying to define SW to do an AX.25 FCS
> calculation, I threw a bit of code together for the Bell 202 modem
> on the DSP-93, so I could push raw packets out the DSP-93 UART
> and get a look at the structure of the FCS. In case anyone is
> interested:
>
>   All bytes in the packet EXCEPT the FCS are sent LSB 1st.
>
>   The FCS is sent high byte first, MSB first. The AX.25 spec nicely
>   gets around defining this by referring to an obscure CCITT spec.

Thanks for the feedback Frank. I have scratched my head over that one as well. I recently have made a hack to Pawel's AX25DRV program, which is a TSR that reads the raw bit stream from the serial port. This program very nicely unscrambles the raw data and provides the interrupt hook for NOS's "attach" command. This way I had a TI320C26 DSK running on HF Packet (of course only reading the mail). This agrees with your next comment.

>
> Based on this quick, rough hack, I don't think there would be any problem
> adding HDLC and KISS to the 300 and 1200 bps modems - if someone has the
> time and interest to pursue it. For the reasons you explained, I don't
> this is practical for 9600 bps.
>
> 73 Frank WB5IPM
>
>

Not to pursue this thread much further here on HFSIG, I just wanted to leave this with a last comment. I wonder if the 9600 baud DSP-93 modem problem could not be solved using a multirate technique, i.e., a number of downsample/upsample filters with the appropiate fractional ratio to get you to an exact multiple of 9600. This might consume less clock cycles and leave the possibilities for other features. Just a thought.

--Johan

75. Re: [HFSIG:917] Sound Blaster

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Wed, 28 Feb 1996 08:59:24 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960228085514.27038B-100000@ucs.orst.edu>
Rendered body type: text/plain

Hi Brian,

Nice to hear from you on the list. I think your experiences with the CL
products at that level basic level is quite interesting.

You certainly are in ahead of the pack with your work - hope a solution
to the sampling problem can be found.

Keep us posted on your activities.

--Johan

76. Re: [HFSIG:917] Sound Blaster

From: Clifford Buttschardt <cbuttsch@biggulp.callamer.com>
Date: Wed, 28 Feb 1996 10:26:52 -0800 (PST)
To: Brian Beezley <k6sti@n2.net>
Cc: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960228102227.23115G-100000@biggulp.callamer.com>
Rendered body type: text/plain

Hi Brian, Johan and the group.. First Brian, I have both of your
E-messages and especially appreciate the comments to hfsig regarding the
sound blaster. I am more than willing to wait until a laptop can be
arranged to do the job. Johans explaination of the difficulties with the
DSP 93 on 9600 baud as well as AMTOR/PACTOR was even more interesting.
Thanks to all. Cliff W6HDO

77. Re: [HFSIG:918] Re: QST tidbits

From: Clifford Buttschardt <cbuttsch@biggulp.callamer.com>
Date: Wed, 28 Feb 1996 13:52:45 -0800 (PST)
To: Robert.Glassey@nmp.nokia.com
Cc: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960228135145.5274E-100000@biggulp.callamer.com>
Rendered body type: text/plain

Your comments regarding BPSK and demods without processor well noted!
Thanks....Cliff Buttschardt   W6HDO

78. Re: [HFSIG:917] Sound Blaster

From: Carol Byers <crlbyers@garlic.com>
Date: Wed, 28 Feb 1996 17:03:30 -0800
To: hfsig@tapr.org
Message-ID: <3134FB62.6E19@garlic.com>
Rendered body type: text/plain

Hey Brian,

When you were real young, did you live on o'Dell street in Edison Park,
Il.??

79. Re: [HFSIG:921] Re: QST tidbits

From: fperkins@onramp.net
Date: Wed, 28 Feb 1996 20:57:43 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <199602290257.UAA26330@mailhost.onramp.net>
Rendered body type: text/plain

Hi Johan,

This (up/down resampling) might be possible. It would be great if the AIO
sampling rate was capable of fine adjustment (corse adjustment is possible but
would "shock" the DSP calculations). With fine adjustment,
you could lock the AIO sampling rate to the locally extacted clock and get good
results with a much lower sampling rate.

73 Frank WB5IPM

80. Re: Latest news

From: N0AOT@aol.com
Date: Thu, 29 Feb 1996 00:57:34 -0500
To: forrerj@ucs.orst.edu
Cc: amsat-bb@amsat.org, hfsig@tapr.org
Message-ID: <960229005733_156262947@emout09.mail.aol.com>
Rendered body type: text/plain

In a message dated 96-02-27 14:47:19 EST, forrerj@ucs.orst.edu (Johan Forrer)
writes:

>The main problem with 9600 baud is that, besides the modem (which is
>quite simple), is data and clock recovery and then HDLC decoding. For

>9600 baud the symbols are very short and ideally one would like to have
>good edge resolution, say 1/250 of the symbol max jitter. That calls for
>very heavy duty front end oversampling. That is why the TAPR DSP-93
>needs an outboard TNC to do this.

>We have been using the Motorola 56002 EVM for this ($150) and have
>managed to do it all on the DSP chip, i.e. modem + HDLC and serial
>communications at 19.2K. There is one fellow that tells me he uses it
>with WISP.

Hi Dr. Johan,

Thanks for the nice note. I had no idea someone already had a 9600 baud modem
running on the EVM! This makes me wonder what else is already available?
Sounds like the EVM is more versatile than the DSP-93... Does anyone keep a
list/library of all the stuff that is currently available for the EVM?

If 9600 baud and amtor/pactor are already available, I'll order one tomorrow!
 Then I can use it to develop new stuff besides... Neat!

In order to recommend a pacsat radio to you, I need to know more about your
requirements. Personally, I am mainly interested in the pacsats for
world-wide email, so the radio requirements and antenna requirements are be
much less than for working ssb and sstv on ao-13, for instance. I have made
minor mods to a Kenwood TM-733a dual bander for pacsat use. These radios are
much cheaper than "real" satellite radios. If you let me know what you want
to do on the satellites, I might be able to help you more...

Thanks Much,

--Bob, n0aot@amsat.org

81. Re: Sound Card Interface

From: Peter Schulze <72253.2602@compuserve.com>
Date: 29 Feb 96 11:03:57 EST
To: HFSIG <hfsig@tapr.org>
Message-ID: <960229160356_72253.2602_EHJ113-3@CompuServe.COM>
Rendered body type: text/plain

Hi Brian,

>Not a bad idea, Peter, except that then I would have to deal with the
>real-time limitations of both the Windows sound interface and the Creative
>Labs device driver it calls. Although cascaded, perhaps these interfaces
>are better suited for low-delay, continuous, real-time I/O. I don't know.
>At least in DOS, you can obtain direct access to the sound card hardware
>without asking permission and then do whatever is necessary to get it to
>work. Apparently due to the demands of game developers, Win95 now
>incorporates a DirectSound interface that lets a program bypass Windows and
>do the same thing.

Directsound won't help you a lot as it is mainly targeted to mix and output
sounds
in sync with screen events. There is nothing in there on the wave input side.

As somebody mentioned before, as long as you are not tight on timing (read RTTY
for example)
you can live with the delays, but when you need to keep in pace with an ARQ
protocol
it can become difficult.
I wonder if anybody ever thought about a new ARQ protocol that works 'one
behind' ?
I mean, that you prepare the 'answer' block while receiving a block and it is
then sent
next time you are on TX. Hope you understand what i mean. This would give ample
time
to analyze a recieved block of data and prepare the reply.

I shall set up some kind of timing loop here one day and see what delays are
involved
when getting wave input under windows with the win drivers... trigger some kind
of signal
on the mic-input under program control and see how long it takes before it
appears
in the samples.

I see a discussion about ways to implement AX25 for the new HF modems.
To my opinion thats a bad track. To much overhead, lack of forward error
correction etc..

It may seem appealing at the first hand to gain easy access to an existing
application
software base, but long time the disadvantages of AX25 will dominate..
Same is true for TCP/IP... just to much overhead for low thruput channels.
Makes no sense to put all effort into faster modems and then eat it up
with protocols that don't fit...

73 Peter

82. HF data link layer (was:Sound Card Interface)

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Thu, 29 Feb 1996 16:58:30 -0800
To: hfsig@tapr.org
Message-ID: <199603010051.QAA08316@PEAK.ORG>
Rendered body type: text/plain

Peter,

>As somebody mentioned before, as long as you are not tight on timing (read RTTY
>for example)
>you can live with the delays, but when you need to keep in pace with an ARQ
>protocol
>it can become difficult.

Yes that is correct. Pactor I for instance has a frame cycle time of 1250
ms, the data being 960 ms, thus the "dead time" is some 290 ms. The "dead
time" is taken up by a) propagation time to the receiver, b) the reverse
channel control command, c) tx/rx changeover delays, d) sending the control
signal. Subtracting the time for sending the control signal, i.e., 120 ms,
leaves 170 ms for all the propagation times and changeover times. This
scheme allows for a 20,000 km maximum distance point-to-point. It is obvious
that any other delays will probably mess things up.

>I wonder if anybody ever thought about a new ARQ protocol that works 'one
>behind' ?
>I mean, that you prepare the 'answer' block while receiving a block and it is
>then sent
>next time you are on TX. Hope you understand what i mean. This would give
ample
>time
>to analyze a recieved block of data and prepare the reply.

There is an ARQ protocol that works like this, i.e., only way down the line
does the receiver request certain blocks to be resent. I think this is
called "Go back N" ARQ. Implementation of these things gets complicated
rather quickly because one have to assume that the reverse channel is also
subject to being corrupted. So there are a lot of shuffling going on for
missing parts. I hope I never have to do one of those :-)

>I see a discussion about ways to implement AX25 for the new HF modems.
>To my opinion thats a bad track. To much overhead, lack of forward error
>correction etc..
>It may seem appealing at the first hand to gain easy access to an existing
>application
>software base, but long time the disadvantages of AX25 will dominate..

I don't think you have to worry much about this Peter. What we have done was
only a quick and dirty hack to try out some modems.

You must have been following the discussions over the past few weeks on the
proposed new data link layer. We were discussing certain parts of the frame,
i.e., the sync vector and the reverse channel signaling. What goes inbetween
these are still undecided. Ideally though I would like to see convolutional
codes with the ability to use punctured codes when conditions are good.

These are only suggestions, so it's open for discussion. Perhaps you would
like to share your ideas??

>Same is true for TCP/IP... just to much overhead for low thruput channels.

I think you mean encapsulating TCP/IP in the existing schemes, which I agree
is not workable. Hopefully there would be a way to minimize the overhead,
yet retain the ability to utilize networking capabilitie with the scheme
proposed above.

Again, we welcome input on this topic.

--Johan

83. Re: Need Recommendation for DSP Design

From: Clifford Buttschardt <cbuttsch@slonet.org>
Date: Thu, 29 Feb 1996 17:02:49 -0800 (PST)
To: David Nulton <dnult@atlas.axiom.net>
Cc: Johan Forrer <forrerj@ucs.orst.edu>, hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960229165312.19842C-100000@biggulp.callamer.com>
Rendered body type: text/plain

Hi David. Thanks for the suggestion on the 320. I did not know that 16 bit processing was used and therefore switching is needed to use 8 bit EPROMS. Good suggestion. I am trying to develop some newer BiPhase Frequency Shift decoding protocols and at the same time, keep it simple enough to where ordinary people can participate. In general, we have been using the computer to maximum ability and making up the difference in hardware. So far we have reached the level where a Pentium is required and have backed down to where 486/66 units will operate. This still seems to exceed the capabilities of the 320. We have gotten this far by "stealth"---using clever programming, but I have no clue if the existing TMS320C25 programs have tried to increase speed in a like manner.
Thanks again for your help. Cliff Buttschardt W6HDO Cal Poly, SLO, CA

March 1996

Original source file: hfsig.9603
Source: https://archive.org/details/tapr-1997
Recovered from: https://archive.org/download/TAPR-HF_1996-03/TAPR-HF_1996-03.txt

1. HF Channel Capacity

From: dona@tridelta.com (Don Adams)
Date: Fri, 1 Mar 1996 10:23:27 -0500 (EST)
To: hfsig@tapr.org
Message-ID: <199603011523.KAA20725@sparcy.tridelta.com>
Rendered body type: text/plain

I've been monitoring tis sig for a few months now and reading the
back issues down loaded from tapr web site. I took a lot of DSP
in college 20 years ago and only now the hardware has finally
caught up. In a previous life I designed Bell 201 modem compatables
and about 3 years ago used an Areil 56000 to proto type some
adaptive control loops.

I've been looking at the 2 various designs that have been around here.
The phase modulated and the multi freq usually with a phase modulation
enhancement.

The real question here is what IS the channel capacity of the HF channel
that we're talking about. It can be looked at in the short term and long
term (ie what is the short term and long term).   What is the advantage
of a multi freq phase moduated signal vs just a single freq more phases
and amplitudes, moduated signal. They both use up the channel states.

Or what about a signal that has several orthoganal signals on it that
could be decoded completely with good signal to noise ratios and only
partially with bad S/N. ie a course structure such as Ur biphase and an
addition fine structure (phase or amplitude or both etc etc) that could
only be resolved if the channel is good. Then using an ACK that
indicates how much was resolved out of the signal.

It all boils down to how many unique states can I resolve at this symbol
period. ie what is the channel capacity?

                                        over and out....   Don
WA8QZZ

---

2. Re: [HFSIG:931] HF Channel Capacity

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Fri, 01 Mar 1996 19:38:51 -0800
To: hfsig@tapr.org
Message-ID: <9603020331.AA29812@ucs.orst.edu>
Rendered body type: text/plain

Hi Don,

Welcome aboard - your experiences sounds very useful. Perhaps you can add to
the ideas that we have been entertaining and experimenting with.

>I've been monitoring tis sig for a few months now and reading the
>back issues down loaded from tapr web site. I took a lot of DSP
>in college 20 years ago and only now the hardware has finally
>caught up. In a previous life I designed Bell 201 modem compatables
>and about 3 years ago used an Areil 56000 to proto type some
>adaptive control loops.
>
>I've been looking at the 2 various designs that have been around here.
>The phase modulated and the multi freq usually with a phase modulation
>enhancement.
>
>The real question here is what IS the channel capacity of the HF channel
>that we're talking about. It can be looked at in the short term and long
>term (ie what is the short term and long term).   What is the advantage
>of a multi freq phase moduated signal vs just a single freq more phases
>and amplitudes, moduated signal. They both use up the channel states.
>
>Or what about a signal that has several orthoganal signals on it that
>could be decoded completely with good signal to noise ratios and only
>partially with bad S/N. ie a course structure such as Ur biphase and an
>addition fine structure (phase or amplitude or both etc etc) that could
>only be resolved if the channel is good. Then using an ACK that

>indicates how much was resolved out of the signal.
>
>It all boils down to how many unique states can I resolve at this symbol
>period. ie what is the channel capacity?
>
>   over and out....   Don
>WA8QZZ
>
>

Good question. I don't have an answer - I can suggest taking Shannon's
channel capacity formula, but keep in mind that is for a channel with
gaussian noise. The HF channel behaves quite different depending at what
frequency you are working, what time of the day it is, which season and
where along the sun-spot cycle you are.

Thus far, we have not yet explored any form of intelligent reverse channel
signaling. It has been suggested by Rolf Sommerhalder and Tom McDermot has
mentioned work done on a single channel 9600 baud modem for HF with dynamic
channel equalization. Tom might be able to tell us a bit more about that
work, but I think it uses training sequences embedded in the data stream
that allows the modem to do channel equalization on the fly while receiving
data.

All interesting stuff - we are interested to hear what you have in mind.

--Johan, KC7WW

3. C5x DSK Assembler

From: "Steven R. Bible" <srbible@mindport.net>
Date: Sat, 02 Mar 1996 15:07:23 -0500
To: hfsig@tapr.org
Message-ID: <1.5.4b11.32.19960302200723.00685a30@mindport.net>
Rendered body type: text/plain

I found a funny thing about the assembler for the C5x DSK. Perhaps someone
could shed some light on this problem.

I downloaded the latest assember from ftp://ftp.ti.com/mirrors/tms320bbs/,
it is entitled "DSK5A110.EXE." It is version 1.10.

Next I downloaded from the same site "5XDSKFIR.EXE" which is a set of FIR
filters (hi, lo, band, and stop pass). Initially these filters have coded
into them a random noise generator. You can comment out the noise generator
and enable the INput RCA jack. I did this and re-assmbled the code with
version 1.10 of the assembler. The new program did not work.

After a lot of fooling around I tried the older assembler, version 1.02.
And sure enough I could assemble the code and it worked. Now I could sample
the INput and hear the effects of the respective filters.

Any idea what is wrong or what I am doing wrong?

Thanks,

 - Steve, N7HPR

srbible@mindport.net
n7hpr@amsat.org
n7hpr@tapr.org

4. Re: [HFSIG:933] C5x DSK Assembler

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Sat, 02 Mar 1996 20:42:20 -0800
To: hfsig@tapr.org
Message-ID: <9603030435.AA07573@ucs.orst.edu>
Rendered body type: text/plain

Hi Steve,

There were two recent postings on COMP.DSP reporting that the latest C5x
assembler was generating bad code. You may want to check that out to see if
those support your findings.

--Johan

>I found a funny thing about the assembler for the C5x DSK. Perhaps someone
>could shed some light on this problem.

>
>I downloaded the latest assember from ftp://ftp.ti.com/mirrors/tms320bbs/,
>it is entitled "DSK5A110.EXE." It is version 1.10.
>
>Next I downloaded from the same site "5XDSKFIR.EXE" which is a set of FIR
>filters (hi, lo, band, and stop pass). Initially these filters have coded
>into them a random noise generator. You can comment out the noise generator
>and enable the INput RCA jack. I did this and re-assmbled the code with
>version 1.10 of the assembler. The new program did not work.
>
>After a lot of fooling around I tried the older assembler, version 1.02.
>And sure enough I could assemble the code and it worked. Now I could sample
>the INput and hear the effects of the respective filters.
>
>Any idea what is wrong or what I am doing wrong?
>
>Thanks,
>
> - Steve, N7HPR
>
>srbible@mindport.net
>n7hpr@amsat.org
>n7hpr@tapr.org
>
>

5. Re: [HFSIG:921] Re: QST tidbits

From: "Jarkko Vuori" <Jarkko.Vuori@hut.fi>
Date: Sun, 3 Mar 1996 19:26:39 +0200
To: hfsig@tapr.org
Message-ID: <199603031726.TAA24916@wooster.hut.fi>
Rendered body type: text/plain

> Not to pursue this thread much further here on HFSIG, I just wanted to
> leave this with a last comment. I wonder if the 9600 baud DSP-93 modem
> problem could not be solved using a multirate technique, i.e., a number
> of downsample/upsample filters with the appropiate fractional ratio to
> get you to an exact multiple of 9600. This might consume less clock
> cycles and leave the possibilities for other features. Just a thought.
>
> --Johan
>
>
>
>

6. Changing the sample rate

From: "Jarkko Vuori" <Jarkko.Vuori@hut.fi>
Date: Sun, 3 Mar 1996 19:50:12 +0200
To: hfsig@tapr.org
Cc: Jarkko.Vuori@hut.fi
Message-ID: <199603031750.TAA24928@wooster.hut.fi>
Rendered body type: text/plain

> Not to pursue this thread much further here on HFSIG, I just wanted to
> leave this with a last comment. I wonder if the 9600 baud DSP-93 modem
> problem could not be solved using a multirate technique, i.e., a number
> of downsample/upsample filters with the appropiate fractional ratio to
> get you to an exact multiple of 9600. This might consume less clock
> cycles and leave the possibilities for other features. Just a thought.

Hi Johan,

Sample rate altering using DSP techniques is not very complex
thing to do at all. You can combine the receive matched filter,
interpolation/decimation filters and symbol timing
filter to one FIR filter with time varying coefficients. This can
be implemented very efficiently with modern DSPs.

Look for example my 9600 bit/s G3RUH modem (g3ruh.asm file).
I have fixed rate sampling at 48 kHz. I have 2/5
interpolator/decimator in order to get the 19200 Hz sampling
rate suitable for the 9600 bit/s decision making. Because this
interpolator/decimator is a simple FIR filter, I can combine
(by convolving the filter coefficients) the receive raised cosine
filter and interpolation/decimation filters together. This
combined filter has the same number of taps than those separate
filters, so there is no performance gain, but this single
combined filter improves the structure of the DSP program.

Jarkko, OH2LNS

7. Re: [HFSIG:933] C5x DSK Assembler

From: dibene@VNET.IBM.COM (Alberto di Bene)
Date: Mon, 4 Mar 96 10:44:58 EST
To: SRBIBLE@MINDPORT.NET
Cc: hfsig@tapr.org
Message-ID: <199603040943.DAA23434@tapr.org>
Rendered body type: text/plain

In a previous message, you wrote:
>I found a funny thing about the assembler for the C5x DSK. Perhaps someone
>could shed some light on this problem.
>
>I downloaded the latest assember from ftp://ftp.ti.com/mirrors/tms320bbs/,
>it is entitled "DSK5A110.EXE." It is version 1.10.
>
>Next I downloaded from the same site "5XDSKFIR.EXE" which is a set of FIR
>filters (hi, lo, band, and stop pass). Initially these filters have coded
>into them a random noise generator. You can comment out the noise generator
>and enable the INput RCA jack. I did this and re-assmbled the code with
>version 1.10 of the assembler. The new program did not work.
>
>After a lot of fooling around I tried the older assembler, version 1.02.
>And sure enough I could assemble the code and it worked. Now I could sample
>the INput and hear the effects of the respective filters.
>
>Any idea what is wrong or what I am doing wrong?
>
>Thanks,
>
> - Steve, N7HPR
>
>srbible@mindport.net
>n7hpr@amsat.org
>n7hpr@tapr.org
>

 Steve, avoid like pest the V 1.10 of the DSK5A assembler. It is full of
 horrendous errors.... I wonder which testing procedures they use before
 releasing a product.. probably none... I have the impression that the
 assembler, loader and debugger delivered with the DSK are an after-tought
 written by a summer student whose main interest was to earn a few dollars
 for the college. I am trying to post to comp.dsp with the results of my
 testing, and I'll do it as soon as the news server will stop telling me
 that it has exhausted the space for new postings.

 As an example :

```
00010 0a06 0812          lamm    PASR    <---- WRONG ! Must assemble as 080a
00011 0a07 0813          lamm    PAER    <---- WRONG ! Must assemble as 080b
00012 0a08 0814          lamm    TREG0   <---- WRONG ! Must assemble as 080c
00013 0a09 0815          lamm    TREG1   <---- WRONG ! Must assemble as 080d
00014 0a0a 0816          lamm    TREG2   <---- WRONG ! Must assemble as 080e
00015 0a0b 0817          lamm    DBMR    <---- WRONG ! Must assemble as 080f
```

 Moreover, if you use the directive .ps xxxx where xxxx is whatever address,
 it simply resets the code space counter to 0a00h, ignoring your xxxx.

 The same for .ds xxxx which resets the data space counter to 0e00h, no
 matter what xxxx you specify.

 Such pieces of cr*p should never be released. The errors are such that even
 a minimal testing would have pinpointed them. It seems that they didn't
 find the time for *any* testing.

 For now it is wiser to stay with V 1.03 - I have been told that a public
 domain assembler for the 'C50 DSK is available for Linux. I am trying to
 get hold of it.

```
Alberto di Bene         dibene@vnet.ibm.com     i2phd@amsat.org
-------------------------------------------------------------------------
     Microsoft gave you windows. IBM gives you the whole house.         )
-------------------------------------------------------------------------
```

8. Re: C5x DSK Assembler

From: hasch@ihf.uni-stuttgart.de (Juergen Hasch)
Date: Mon, 4 Mar 1996 13:51:21 +0100
To: hfsig@tapr.org
Message-ID: <9603041251.AA16206@ihfibm1.ihf.uni-stuttgart.de>
Rendered body type: text/plain

You wrote:

> For now it is wiser to stay with V 1.03 - I have been told that a public
> domain assembler for the 'C50 DSK is available for Linux. I am trying to
> get hold of it.
>
>Alberto di Bene   dibene@vnet.ibm.com   i2phd@amsat.org

You could try 'AS' from Alfred Arnold, which available at
'ftp.uni-stuttgart.de/pub/systems/pc/lang/as' and has (among many others) code
generators for the 320C2x and 320c5x chips. The TMS320 code generators were
written by Tom HB9JNX.

AS runs with DOS and OS/2, the Pascal source is available if you want to write
your own code generator. Documentaion is only in german right now.

I have written all programs for my 320C25 DSP card with AS and never had any
problems, AS is definitely much better than any of the crippled DSK assemblers
from TI.

Juergen Hasch DG1SCR

Institut fuer Hochfrequenztechnik	hasch@ihf.uni-stuttgart.de
Universitaet Stuttgart	
Germany	Packet Radio: dg1scr@db0rbs.#bw.deu.eu

9. 1996 ARRL and TAPR Digital Communications Conference

From: Greg Jones <wd5ivd@tapr.org>
Date: Mon, 4 Mar 1996 15:52:53 -0600 (CST)
To: netsig@tapr.org (NETSIG mailing), bbssig@tapr.org (BBS SIG mailing),
aprssig@tapr.org (BBS SIG mailing), hfsig@tapr.org (HF SIG mailing),
dsp-93@tapr.org (DSP-93 Build), ss@tapr.org, tapr-tnc@tapr.org
Message-ID: <199603042152.PAA21342@tapr.org>
Rendered body type: text/plain

Press Release. Please distribute as widely as possible. http://www.tapr.org
March 1st, 1996

ARRL and TAPR Digital Communications Conference
September 20-22, 1996
Seattle, WA, U.S.A.

1996 ARRL and TAPR Digital Communications Conference

Mark your calendar and start making plans to attend the year's premier event
in amateur radio digital communications --

The 1996 ARRL and TAPR Digital Communications Conference will be held
September 20-22, 1996, in Seattle, Washington -- just minutes from the
SeaTac Airport. Local sponsors include the Puget Sound Amateur Radio TCP/IP
Group and Boeing Employees Amateur Radio Society (BEARS).

The ARRL and TAPR Digital Communications Conference is an international
forum for radio amateurs and experts in digital communications, networking,
and related technologies to meet, publish their work, and present new ideas
and techniques for discussion. Presenters and attendees will have the
opportunity to exchange ideas and learn about recent hardware and software
advances, theories, experimental results, and practical applications. This
year marks the first year in which the ARRL Digital Communications
Conference and TAPR Annual General Meeting have joined into one conference!

Anyone interested in digital communications is invited to submit a paper for
publication in the Conference Proceedings. Presentation at the Conference
is not required for publication. Papers are due by July 23, 1996, and
should be submitted to Maty Weinberg, ARRL, 225 Main St., Newington, CT
06111 U.S.A. or via Internet at lweinberg@arrl.org. Please contact Maty for
detailed format requirements. Papers are due by July 23, 1996

ARRL and TAPR especially welcome papers from full time students to compete
for the first annual student papers award. Two $500 travel awards will be

given, one in each of the following categories: a) best technical/theory-oriented paper by a student, and b) best educational or community-oriented application paper by a student. The paper should relate directly to a wireless digital communication topic (see guidelines for more information). Papers coauthored by educators or telecommunications professionals are also eligible for this award, as long as a student is the first author. First year awards have been funded through a grant by The ARRL Foundation, Inc. Deadline for receipt of finished student paper manuscript: June 11, 1996. Please note that this deadline is different than the general conference submission date. For full details and paper guidelines, contact TAPR or check http://www.tapr.org.

The current schedule has a workshop and student papers on Friday afternoon. Conference Papers, Introductory thread, Lunch, and Dinner on Saturday. Sunday will have additional paper sessions, if necessary, as well as one or more workshops. Workshops will include Keith Sproul, WU2Z, on APRS packet-location software, Dewayne Hendricks, WA8DZP , on "How to utilize Part 15 Radios for Ham Applications" and a workshop on Wireless Networking using WA4DSY 56K RF modem technology and its accessories. See the advanced registration form for more details as well as http://www.tapr.org

For more information on the Conference, registration, and hotel reservations please contact TAPR at the address below.

TAPR Office:
8987-309 E. Tanque Verde Rd. #337
Tucson, AZ 85749-9399
Phone: (817) 383-0000
Fax: (817) 566-2544

Internet E-Mail: tapr@tapr.org
Internet Web: http://www.tapr.org

10. TEST ONLY!

From: Fermax.esp@vlc.servicom.es
Date: Tue, 05 Mar 1996 18:51:16 +0100
To: hfsig@tapr.org
Message-ID: <199603051756.SAA00058@ns.mad.servicom.es>
Rendered body type: text/plain

JOSE V. GAVILA
R&D Engineer
FERMAX ELECTRONICA S.A.E.

11. Re: [HFSIG:938] Re: C5x DSK Assembler

From: Jim Sanford <JSANFORD@infi.net>
Date: Mon, 04 Mar 1996 19:06:39 -0500
To: hfsig@tapr.org
Message-ID: <199603050007.TAA21459@mh004.infi.net>
Rendered body type: text/plain

Jurgen: Any mirror sites? Couldn't get to yours. Thanks & 73, Jim
wb4gcs@amsat.org
>You could try 'AS' from Alfred Arnold, which available at
>'ftp.uni-stuttgart.de/pub/systems/pc/lang/as' and has (among many others) code
>

12. 56002 EVM Documentation

From: N0AOT@aol.com
Date: Tue, 5 Mar 1996 19:39:51 -0500
To: hfsig@tapr.org
Message-ID: <960305193950_238470400@emout10.mail.aol.com>
Rendered body type: text/plain

Hi All,

I ordered some the on the E56002 VM from Motorola's Web site:
http://www.design-net.com/dsp/DSPhome.html, and I received the package via
UPS in about 2 days!

Now that's what I call service! Free too...

--Bob Carlson, n0aot@amsat.org

13. Re: C5x DSK Assembler

From: hasch@ihf.uni-stuttgart.de (Juergen Hasch)
Date: Wed, 06 Mar 96 09:57:47 CET

To: hfsig@tapr.org
Message-ID: <9603060855.AA20973@ihfrisc.ihf.uni-stuttgart.de>
Rendered body type: text/plain

On Tue, 5 Mar 1996 12:20:33 -0600 (CST) you wrote:

>Jurgen: Any mirror sites? Couldn't get to yours. Thanks & 73, Jim
>wb4gcs@amsat.org

Hello Jim,

since the recent version of AS with C5x support is brand new, I don't believe
there are any other sites (at least outside of europe) yet.
If you really can't get through, I can send you AS by Email (0.5 Meg!).

73's de Juergen DG1SCR
                      hasch@ihf.uni-stuttgart.de

14. AS and macros

From: hasch@ihf.uni-stuttgart.de (Juergen Hasch)
Date: Wed, 06 Mar 96 09:59:34 CET
To: hfsig@tapr.org
Message-ID: <9603060855.AA31722@ihfrisc.ihf.uni-stuttgart.de>
Rendered body type: text/plain

Hello Johan,

porting macros to AS should not be too difficult.

This is what the AS documentation says:
  The same conventions as for symbols also apply to macro parameters, with the
  exception that no points and underscores are allowed. This restriction
  is neccessary because the underscore allows to concatenate macro parameter
  names to a single symbol, as shown in the following example:

```
concat  macro   part1,part2
        call    part1_part2
        endm
```

If you call the macro

```
        concat  Modul,Funktion
```

You will get

```
        call    Modul_Funktion
```

If you want to port Alexander's HFSIM for example:

```
F0_BUF1   equ  0400h
 gets
F0_BUF_1  equ  0400h
```

 and

```
OVS0TP  $MACRO  TP_NR
        TP      F0_BUF:B_NR.S:,TP:TP_NR:

        LDPK    F0_OUT_P
        SACH    F0_OUT:B_NR.S:,3

        B       END0_:B_NR.S:_TP2
        $ENDM
```

 gets

```
OVS0TP        MACRO        TPNR,BNR
        TP      F0_BUF_BNR,TP_TPNR

        LDPK    F0_OUT_P
        SACH    F0_OUT_BNR,3

        B       END0_BNR_TP2
        ENDM
```

73's de Juergen DG1SCR
                hasch@ihf.uni-stuttgart.de

15. Re: [HFSIG:912] Re: Sync Vectors for HF Digital

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 6 Mar 1996 01:10:51 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199603060910.BAA10645@servo.qualcomm.com>
Rendered body type: text/plain

>For the same argument, I assume that reverse channel signaling for an
>ARQ system needs the same considerations as that is also an unprotected
>sequence?

I see no need for a special reverse channel (in the AMTOR sense) in a
well-coded system. With appropriate FEC you can lengthen your
transmissions considerably and send acks as ordinary frames. Not only
would this avoid a lot of turnaround overhead, but it would be a lot
easier on transmitter relays.

Phil

16. Re: [HFSIG:944] AS and macros

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Wed, 06 Mar 1996 09:21:23 -0800
To: hfsig@tapr.org
Message-ID: <9603061713.AA02264@ucs.orst.edu>
Rendered body type: text/plain

Hello Juergen,

That sounds exactly what I needed - I am in the process of porting
Alexander's simulator to a TI DSK for my own testing purposes. I did not use
the "_" concatenator construct, so that is why my efforts failed. Everything
else seems to work out OK.

The other bit that is missing - what is the output format of the P2HEX and
P2BIN programs? Seems like a good way to generate a compacted version for
loading into the DSP. Do you perhaps know of a C25/6 code loader that will
read such a file?

Thanks a lot for your help.

--Johan

17. Re: [HFSIG:945] Re: Sync Vectors for HF Digital

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Wed, 06 Mar 1996 09:21:26 -0800
To: hfsig@tapr.org
Message-ID: <9603061713.AA20374@ucs.orst.edu>
Rendered body type: text/plain

Hi Phil,

>I see no need for a special reverse channel (in the AMTOR sense) in a
>well-coded system. With appropriate FEC you can lengthen your
>transmissions considerably and send acks as ordinary frames. Not only
>would this avoid a lot of turnaround overhead, but it would be a lot
>easier on transmitter relays.
>
>Phil
>
>

My casual observations on monitoring HF packet transmissions is that these
ACKs sent as ordinary frames is one of the reasons for the inefficiency of
HF packet. A much shortened frame may be workable, but then we are talking
about variable frame lengths which needs some further thought. For example,
if we know where a frame starts, how do we know where it ends when its
length could be anything in the new scheme of things.

I guess I would need to see the arguments as to why the reverse channel
signaling should use ordinary frames, especially in an HF environnment.

I appreciate your further thoughts on this.

--Johan

18. Re: AS and macros

From: hasch@ihf.uni-stuttgart.de (Juergen Hasch)
Date: Wed, 06 Mar 96 21:17:32 CET
To: hfsig@tapr.org
Message-ID: <9603062012.AA07025@ihfrisc.ihf.uni-stuttgart.de>
Rendered body type: text/plain

Hello Johan,

On Wed, 6 Mar 1996 11:28:39 -0600 (CST) you wrote:

>
>That sounds exactly what I needed - I am in the process of porting
>Alexander's simulator to a TI DSK for my own testing purposes. I did not use
>the "_" concatenator construct, so that is why my efforts failed. Everything
>else seems to work out OK.
>
Alexander sent me his latest version of the simulator just today and I have
made all neccessary changes to get it to run with AS. If you want to save
some work I can send you the changed source code.
Except the macro stuff all other changes are minor, just a few illegal commands
the DSK assembler didn't notice :-)
As soon as I get an 320C26 processor, I will port the simulator to my own DSP
card which currently runs with a 320C25. The C26 is needed because the simulator
uses all of the internal RAM.

>The other bit that is missing - what is the output format of the P2HEX and
>P2BIN programs? Seems like a good way to generate a compacted version for
>loading into the DSP. Do you perhaps know of a C25/6 code loader that will
>read such a file?
The *.P output format is quite nice, it even contains a processor ID byte.
I can send you an example of how the *.P-files download was done for my DSP
card, this should help you using P-files.
You could also ask Tom HB9JNX. He should have something for the DSK, as he
uses AS for his DSP programs.

>
>Thanks a lot for your help.
Glad I could help.

73's de Juergen DG1SCR, hasch@ihf.uni-stuttgart.de
PS:
You can get AS to print english messages if you download the AS pascal source
from the ftp-server in stuttgart (ftp.uni-stuttgart.de) and recompile it with the english message
resources contained in 'aseng.zip' (only the slightly older AS1412 source is there, but it works for the
 320C26).

19. Re: [HFSIG:947] Re: Sync Vectors for HF Digital

From: chbrain@dircon.co.uk (Charles Brain)
Date: Wed, 06 Mar 1996 19:44:31 +0000
To: hfsig@tapr.org
Message-ID: <199603061946.AA27458@felix.dircon.co.uk>
Rendered body type: text/plain

>My casual observations on monitoring HF packet transmissions is that these
>ACKs sent as ordinary frames is one of the reasons for the inefficiency of
>HF packet. A much shortened frame may be workable, but then we are talking
>about variable frame lengths which needs some further thought. For example,
>if we know where a frame starts, how do we know where it ends when its
>length could be anything in the new scheme of things.
>
>I guess I would need to see the arguments as to why the reverse channel
>signaling should use ordinary frames, especially in an HF environnment.
>
>I appreciate your further thoughts on this.
>
>--Johan
>
>
I believe that the two major problems of H.F packet are a collision sensing
method that is next to useless and the fact that there is no FEC.
I assume we are not trying to use CSMA to share the channel. Either
because it is running an ARQ where one station is acting as a master or
a slotted response system ?

Have you seen the 1052 Data Link protocol ? There is a copy in postscript
format at

ftp://atanasoff.nmsu.edu/pub/hf/pubs/

I think its called HFDLP or something like it.

Regards Charles G4GUO

20. Re: [HFSIG:947] Re: Sync Vectors for HF Digital

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 6 Mar 1996 13:54:57 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199603062154.NAA00549@qualcomm.com>
Rendered body type: text/plain

At 11:31 AM 3/6/96 -0600, you wrote:

>My casual observations on monitoring HF packet transmissions is that these
>ACKs sent as ordinary frames is one of the reasons for the inefficiency of
>HF packet. A much shortened frame may be workable, but then we are talking
>about variable frame lengths which needs some further thought. For example,
>if we know where a frame starts, how do we know where it ends when its
>length could be anything in the new scheme of things.
>
>I guess I would need to see the arguments as to why the reverse channel
>signaling should use ordinary frames, especially in an HF environnment.
>
>I appreciate your further thoughts on this.

Certainly. Why is HF packet any different from VHF/UHF packet? The channel characteristics, of course. If you can compensate for those characteristics with appropriate coding, then conventional protocols should work as well over HF as they do over VHF/UHF. This is not to say they'll work *well*, of course, because they also have problems on VHF/UHF. :-)

I believe the approaches I discussed in my paper "Toward New Link Layer Protocols" are as appropriate for HF as for anywhere else. In there I describe a packet format that consists of a fixed-size header followed by a variable length data segment. The header is always heavily coded, while the data is coded at an effectively varying rate with Type II ARQ-FEC running the link.

When you code effectively, large packets have the advantage of spanning fades, which lets you apply interleaving to the problem. One of the things I've learned from Qualcomm CDMA is that interleaving is a very powerful tool against fading, but we are limited in how much of it we can do because of our tight delay constraints. So we interleave only over a 20ms frame span. If you can relax your delay constraints, and you can for HF data, then you can interleave over much longer spans.

Phil

21. Re: [HFSIG:944] AS and macros

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Wed, 06 Mar 1996 18:07:09 -0800
To: hfsig@tapr.org
Cc: hasch@ihf.uni-stuttgart.de (Juergen Hasch)
Message-ID: <199603070159.RAA01937@PEAK.ORG>
Rendered body type: text/plain

Hello Juergen,

I have now succeeded in assembling Alexander's HF channel simulator for the 320C25.
One small problem that you may be able to help with:

When I generate Intel Hex format output with the P2HEX program, data bytes comes out swapped, i.e., the MSB and LSB order is swapped. Address and line checksums are correct though. Short of writing my own version of P2HEX, do you know of a way to get it right? I have played with the "-m" switch set to 0,1,2 and none are completely right. What am I missing?

With that little problem out of the way, I can load the code into the DSP-93 and actually try it.

73's

--Johan

22. Re: [HFSIG:950] Re: Sync Vectors for HF Digital

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Wed, 06 Mar 1996 18:33:15 -0800
To: hfsig@tapr.org
Message-ID: <199603070225.SAA05878@PEAK.ORG>

Rendered body type: text/plain

Hi Phil,

>
>Certainly. Why is HF packet any different from VHF/UHF packet? The channel
>characteristics, of course. If you can compensate for those characteristics
with
>appropriate coding, then conventional protocols should work as well over HF
>as they do over VHF/UHF. This is not to say they'll work *well*, of course,
because
>they also have problems on VHF/UHF. :-)
>
>I believe the approaches I discussed in my paper "Toward New Link Layer
Protocols"
>are as appropriate for HF as for anywhere else. In there I describe a
packet format
>that consists of a fixed-size header followed by a variable length data
segment.
>The header is always heavily coded, while the data is coded at an effectively
>varying rate with Type II ARQ-FEC running the link.
>
>When you code effectively, large packets have the advantage of spanning
fades, which
>lets you apply interleaving to the problem. One of the things I've learned from
>Qualcomm CDMA is that interleaving is a very powerful tool against fading, but
>we are limited in how much of it we can do because of our tight delay
constraints.
>So we interleave only over a 20ms frame span. If you can relax your delay
constraints,
>and you can for HF data, then you can interleave over much longer spans.
>
>Phil
>
>

If I follow the argument correctly: instead of looking for the ACK in a
fixed time slot, like AMTOR/Pactor ARQ, one now look for a frame header that
can arrive at any time. The protocol probably allows for channel sharing if
you can sort out amongst all these packets. There is no question about the
importance of interleaving.

The argument is thus whether a powerful encoded ACK within an allocated time
slot works as well as decoding a new frame. If the ACK is as good as the
frame sync vector, it's like a new frame with a message of length zero.
Is'nt it? I think the incentive is to move traffic by the least amount of
unncessary overhead by hogging the frequency for a shorter time duration. I
think that is what AMTOR/Pactor ARQ does. The tradeoff is perhaps the gains
by allowing channel sharing.

Which is better?

--Johan

23. Re: [HFSIG:948] Re: AS and macros

From: dibene@VNET.IBM.COM (Alberto di Bene)
Date: Thu, 7 Mar 96 17:30:26 EST
To: HASCH@IHF.UNI-STUTTGART.DE
Cc: hfsig@tapr.org
Message-ID: <199603071628.KAA02079@tapr.org>
Rendered body type: text/plain

In a previous message, you wrote:
>...

>You can get AS to print english messages if you download the AS pascal source
>from the ftp-server in stuttgart (ftp.uni-stuttgart.de) and recompile it with
>the english message resources contained in 'aseng.zip'

>...

I downloaded the ASSRC.ZIP from the Stuttgart ftp site, but it seems to be
relative to version 1412, not 1413. To assemble for the TMS320C50 the
latest version is needed. Do you know if there is any hope to have the source
code for version 1413 placed on that ftp server ? I am afraid that reading
error messages in German would lead me to still more errors :-)

Thanks,

```
Alberto di Bene         dibene@vnet.ibm.com     i2phd@amsat.org
------------------------------------------------------------------------
     Microsoft gave you windows. OS/2 gives you the whole house.        )
------------------------------------------------------------------------
```

24. UPLOAD of the AS assembler

From: Frode Weierud <frode@dxcern.cern.ch>
Date: Thu, 7 Mar 1996 18:25:00 +0100 (MET)
To: HFSIG TAPR <hfsig@tapr.org>
Message-ID: <Pine.ULT.3.91.960307181916.21481A-100000@dxcern.cern.ch>
Rendered body type: text/plain

I have just uploaded Alfred Arnold's AS assembler to the TAPR
FTP server: ftp.tapr.org in the directory: tapr/SIG/hfsig/upload

Below is the contents of the as1413.txt file.

73's Frode, LA2RL

The three archive files as1413.zip, asx1413.zip and asps1413.zip contains
Alfred Arnold's AS assembler.

The archive file as1413.zip contains the basic AS assembler and all its
support files. This executeable is running in Real-Mode.

The archive file asx1413.zip contains a Protected-Mode version of
the AS assembler.

The archive file asps1413.zip contains the full documentation in
Postscript, but unfortunately in German only.

Please find a copy of the readme.txt that accompanies the software.

73's Frode, LA2RL

```
        Frode Weierud                   Phone   : 41 22 7674794
        CERN, SL                        Fax     : 41 22 7679185
        CH-1211 Geneva 23, Switzerland  E-mail  : frode@dxcern.cern.ch
```

Alfred Arnold
Broeltalstr.5a

D-53773 Hennef(Sieg)

E-Mail: a.arnold@kfa-juelich.de

In diesem Verzeichnis befinden sich die Dateien zum
Makroassembler AS. Bei allen Dateien handelt es sich um mit
ZIP komprimierte Archive, die man mit UNZIP oder PKUNZIP wieder
"entfalten" kann. Die Dateien haben folgenden Inhalt
(wobei xxx und y fuer Versionsnummer und Release stehen, die sich ab
und zu aendern ;-)) :

- ASxxxy.ZIP   : Basisdatei. Enthaelt die Real-Mode-Version des
                  Assemblers sowie die Hilfsprogramme und Demos.
                  Diese Datei wird auf jeden Fall benoetigt.
- ASXxxxy.ZIP   : Protected-Mode-Version des Assemblers. Diese
                  enthaelt neben dem Assembler nur die Runtime-
                  Libraries, die Hilfsprogramme aus der ersten
                  Datei sind zusaetzlich erforderlich!!!
- AS2xxxy.ZIP   : 16-Bit OS/2-Version des Assemblers. Hier sind auch
                  die Hilfsprogramme als OS/2-Versionen dabei, jedoch
                  nicht eventuell benoetigte Includes (STDDEFxx.INC)
                  oder Demoprogramme.
- ASTXxxxy.ZIP : LaTeX-Quellen der Dokumentation zu AS, ASX und AS2.
                  Diese Texte finden sich in ASCII-Form in den
                  ersten drei Dateien wieder.
- ASPSxxxy.ZIP : Die Dokumentation in PostScript-Dateien

Es werden folgende Zielprozessoren unterstuetzt:

 - Motorola 68000..68030,683xx inkl. Koprozessor und MMU
 - Motorola DSP56000
 - Motorola/IBM MPC601/MPC505/PPC403

- Motorola 6800, 6805, 6809, 68(HC)11, 68HC16 sowie Hitachi 6301
- Hitachi 6309
- Hitachi H8/300(L/H)
- Hitachi H8/500
- Hitachi SH7000/7600
- Rockwell 6502 und 65(S)C02
- CMD 65816
- Mitsubishi MELPS-740
- Mitsubishi MELPS-7700
- Mitsubishi MELPS-4500
- Mitsubishi M16
- Intel MCS-48/41
- Intel MCS-51
- Intel MCS-96
- Intel 8080/8085
- AMD 29K
- Siemens 80C166/167
- Zilog Z80, Z180, Z380
- Zilog Z8
- Toshiba TLCS-900(L)
- Toshiba TLCS-90
- Toshiba TLCS-870
- Toshiba TLCS-47
- Toshiba TLCS-9000
- Microchip PIC16C54..16C57
- Microchip PIC16C84/PIC16C64
- Microchip PIC17C42
- SGS-Thomson ST62xx
- SGS-Thomson 6804
- Texas Instruments TMS32010/32015
- Texas Instruments TMS3202x
- Texas Instruments TMS320C3x
- Texas Instruments TMS320C5x
- Texas Instruments TMS370xxx
- NEC uPD 78(C)1x
- NEC uPD 75xxx (alias 75K0)

Viel Spass beim Assemblieren! (Fragen und Fehlerberichte bitte
an obige E-Mail-Adresse)

---

25. RE: [HFSIG:928] Re: Sound Card Interface

From: "Sadowski,P O4 USAF JCSAR/EDD" <sadowskp@jcsar.nellis.af.mil>
Date: Thu, 07 Mar 96 09:31:00 cst
To: hfsig <hfsig@tapr.org>
Message-ID: <313F034F@mailgtwy.nellis.af.mil>
Rendered body type: text/plain

Saw this particular comment and I couldn't help but throw in my 2 cents.
US DoD went through extreme pain and agony over the overhead of
the new protocols over low bandwidth channels. After spending some
serious money with many different experts... they came up with a low
overhead protocol... with FEC option... called TACO 2 (tactical comm
protocol). I suggest the SIG look into this protocol.... DoD has been
seeking IEEE or ISO standard for this protocol. Its tailor made for
low BW RF links...

ALOHA
Al   AH6LS

From: hfsig
To: hfsig
Subject: [HFSIG:928] Re: Sound Card Interface
Date: Thursday, February 29, 1996 10:59AM

I see a discussion about ways to implement AX25 for the new HF modems.
To my opinion thats a bad track. To much overhead, lack of forward error
correction etc..
It may seem appealing at the first hand to gain easy access to an existing
application
software base, but long time the disadvantages of AX25 will dominate..
Same is true for TCP/IP... just to much overhead for low thruput channels.
Makes no sense to put all effort into faster modems and then eat it up
with protocols that don't fit...

73 Peter

26. Re: AS and macros

From: hasch@ihf.uni-stuttgart.de (Juergen Hasch)
Date: Thu, 07 Mar 96 20:40:25 CET
To: hfsig@tapr.org
Message-ID: <9603071954.AA25202@ihfrisc.ihf.uni-stuttgart.de>
Rendered body type: text/plain

Hello Johan,

On Wed, 6 Mar 1996 20:00:19 -0600 (CST) you wrote:

>When I generate Intel Hex format output with the P2HEX program, data bytes
>comes out swapped, i.e., the MSB and LSB order is swapped. Address and line
>checksums are correct though. Short of writing my own version of P2HEX, do
>you know of a way to get it right? I have played with the "-m" switch set to
>0,1,2 and none are completely right. What am I missing?

By using '-m 1' you can swap LSB and MSB, isn't this what you want ?
There is a switch '-F DSK' to produce TI DSK output, maybe this is
what you need.
I have to admit I never used anything else than the original P format, so I never
cared much about P2HEX.

>
>With that little problem out of the way, I can load the code into the DSP-93
>and actually try it.
Hopefully you are successfull and get the simulator running. I would be glad to
hear about your experiences with the simulator.

73's de Juergen DG1SCR

27. Re: [HFSIG:955] Re: Sound Card Interface

From: Clifford Buttschardt <cbuttsch@slonet.org>
Date: Thu, 7 Mar 1996 12:16:12 -0800 (PST)
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960307121503.18375F-100000@biggulp.callamer.com>
Rendered body type: text/plain

Now here is the first sensible suggestion in many tries. This protocol
has been around for some time, but we seem to want to reinvent the wheel
each and every time. 72 Cliff Buttschardt W6HDO

On Thu, 7 Mar 1996, Sadowski,P O4 USAF JCSAR/EDD wrote:

>
> Saw this particular comment and I couldn't help but throw in my 2 cents.
> US DoD went through extreme pain and agony over the overhead of
> the new protocols over low bandwidth channels. After spending some
> serious money with many different experts... they came up with a low
> overhead protocol... with FEC option... called TACO 2 (tactical comm
> protocol). I suggest the SIG look into this protocol.... DoD has been
> seeking IEEE or ISO standard for this protocol. Its tailor made for
> low BW RF links...
>
> ALOHA
> Al   AH6LS
>
> ----------
> From: hfsig
> To: hfsig
> Subject: [HFSIG:928] Re: Sound Card Interface
> Date: Thursday, February 29, 1996 10:59AM
>
> I see a discussion about ways to implement AX25 for the new HF modems.
> To my opinion thats a bad track. To much overhead, lack of forward error
> correction etc..
> It may seem appealing at the first hand to gain easy access to an existing
> application
> software base, but long time the disadvantages of AX25 will dominate..
> Same is true for TCP/IP... just to much overhead for low thruput channels.
> Makes no sense to put all effort into faster modems and then eat it up
> with protocols that don't fit...
>
> 73 Peter
>
>

28. Re: [HFSIG:956] Re: AS and macros

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Thu, 07 Mar 1996 14:01:11 -0800
To: hfsig@tapr.org
Message-ID: <9603072153.AA17529@ucs.orst.edu>
Rendered body type: text/plain

Hi Juergen,

Thanks for the advice. The -m 1 option does swap the bytes, but it does something strange with the address fields too. I solved the problem by writing a quicky program that takes the hex output file and writes it in the swapped form. This fixed it and now code loads into the DSP-93.

I will have to relocate the simulator's memory to run on the DSP-93 - my C26 DSK only has expanded P memory, and the simulator needs some stack space that is allocated to grown down from 1500h. The DSP-93 has plenty of P and Data memory, but it is a C25, so the reason for moving data memory.

I would be very interested in working from your modified version of the simulator code, if that is possible. Otherwise I'll work on the previous version - is there some major change between these versions?

Otherwise I discovered a "feature" in the DSP-93 code. It is assumed that "BANZ label" implies "BANZ label,*-" I think this is an old leftover from 320C10 days? It took a bit of puzzling to find that this is the reason why "AS" -generated code would not run on the DSP-93. Anyways all is well now with using AS. I like it a lot.

I'll let you know when I get the simulator code working. Looking forward to receiving the updated version of the simulator code.

73's

--Johan

29. Re: [HFSIG:955] Re: Sound Card Interface

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Thu, 07 Mar 1996 14:01:15 -0800
To: hfsig@tapr.org
Message-ID: <9603072153.AA20548@ucs.orst.edu>
Rendered body type: text/plain

Hi Al,

Any pointers where we can find a description of TACO 2 ?

--Johan

>
>Saw this particular comment and I couldn't help but throw in my 2 cents.
>US DoD went through extreme pain and agony over the overhead of
>the new protocols over low bandwidth channels. After spending some
>serious money with many different experts... they came up with a low
>overhead protocol... with FEC option... called TACO 2 (tactical comm
>protocol). I suggest the SIG look into this protocol.... DoD has been
>seeking IEEE or ISO standard for this protocol. Its tailor made for
>low BW RF links...
>
>ALOHA
>Al   AH6LS
>

30. Re: [HFSIG:959]

From: Marek Kotas <mkotas@ccmail.crosscomm.com>
Date: Fri, 08 Mar 96 04:14:40 EST
To: hfsig@tapr.org
Message-ID: <9603080414.A06220@ccmail.crosscomm.com>
Rendered body type: text/plain

Hello everyone,

General discussion on TACO 2 could be found at:
http://www.tasc.com/NITFS/briefings/TACO2_intro.html

Standard is provided at:
http://www.itsi.disa.mil/ismc/misc/ftpage.html (+ notice of change)

Unfortunately these are WordPerfect files - could somebody convert them to
PostScript or Acrobat, please?

Greetings,
Marek, SP2OVR

_____ Reply Separator _____
Subject: [HFSIG:959] Re: Sound Card Interface
Author: forrerj@ucs.orst.edu (Johan Forrer) at inet
Date: 3/7/96 4:12 PM

Hi Al,

Any pointers where we can find a description of TACO 2 ?

--Johan

>
>Saw this particular comment and I couldn't help but throw in my 2 cents.
>US DoD went through extreme pain and agony over the overhead of
>the new protocols over low bandwidth channels. After spending some
>serious money with many different experts... they came up with a low
>overhead protocol... with FEC option... called TACO 2 (tactical comm
>protocol). I suggest the SIG look into this protocol.... DoD has been
>seeking IEEE or ISO standard for this protocol. Its tailor made for
>low BW RF links...
>
>ALOHA
>Al   AH6LS
>

31. Re: [HFSIG:960] Re:

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Fri, 08 Mar 1996 09:29:35 -0800
To: hfsig@tapr.org
Message-ID: <9603081721.AA15101@ucs.orst.edu>
Rendered body type: text/plain

Thanks Marek,

I got the TACO2 documentation now and will have a loot at it. It seems
primarily for image transmission?

--Johan

>
>Hello everyone,
>
>General discussion on TACO 2 could be found at:
>http://www.tasc.com/NITFS/briefings/TACO2_intro.html
>
>Standard is provided at:
>http://www.itsi.disa.mil/ismc/misc/ftpage.html (+ notice of change)
>
>Unfortunately these are WordPerfect files - could somebody convert them to
>PostScript or Acrobat, please?
>
>Greetings,
>Marek, SP2OVR
>
>_____ Reply Separator
_____
>Subject: [HFSIG:959] Re: Sound Card Interface
>Author: forrerj@ucs.orst.edu (Johan Forrer) at inet
>Date: 3/7/96 4:12 PM
>
>
>Hi Al,
>
>Any pointers where we can find a description of TACO 2 ?
>
>--Johan
>
>>
>>Saw this particular comment and I couldn't help but throw in my 2 cents.
>>US DoD went through extreme pain and agony over the overhead of
>>the new protocols over low bandwidth channels. After spending some

>>serious money with many different experts... they came up with a low
>>overhead protocol... with FEC option... called TACO 2 (tactical comm
>>protocol). I suggest the SIG look into this protocol.... DoD has been
>>seeking IEEE or ISO standard for this protocol. Its tailor made for
>>low BW RF links...
>>
>>ALOHA
>>Al   AH6LS
>>
>
>

32. As again...

From: hasch@ihf.uni-stuttgart.de (Juergen Hasch)
Date: Fri, 8 Mar 1996 19:53:34 +0100 (MEZ)
To: hfsig@tapr.org
Message-ID: <9603081853.AA18987@ihfrs5.ihf.uni-stuttgart.de>
Rendered body type: text/plain

Hello Johan,

> I will have to relocate the simulator's memory to run on the DSP-93 - my C26
> DSK only has expanded P memory, and the simulator needs some stack space
> that is allocated to grown down from 1500h. The DSP-93 has plenty of P and
> Data memory, but it is a C25, so the reason for moving data memory.
If you use a C25 processor, you will have to change some of the code,
because MACD won't move any data in the external data memory (->DMOV).
I have the same problem here and will try to get a C26 sample processor.
(Where can you buy a single CPU...)

>I would be very interested in working from your modified version of the
>simulator code, if that is possible. Otherwise I'll work on the previous
>version - is there some major change between these versions?
Alexander added input signal power measurement and changed the memory
layout. All data now fits in the internal RAM of the C26, you will still
need some external program memory though.

>Otherwise I discovered a "feature" in the DSP-93 code. It is assumed that
>"BANZ label" implies "BANZ label,*-" I think this is an old leftover from
>320C10 days? It took a bit of puzzling to find that this is the reason why
>"AS" -generated code would not run on the DSP-93. Anyways all is well now
>with using AS. I like it a lot.
This is exactly what my data book says...

When I started writing C25 programs, I tried a number of different assemblers,
but I never found anything better then AS.
Btw, the author promised me to compile a AS version with english messages, I
will upload it to the TAPR ftp server on monday, if this is OK.

73's de Juergen DG1SCR

33. Re: [HFSIG:955] Re: Sound Card Interface

From: Clifford Buttschardt <cbuttsch@slonet.org>
Date: Sun, 10 Mar 1996 10:01:31 -0800 (PST)
To: kc7ww@tapr.org
Cc: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960310095759.4485A-100000@biggulp.callamer.com>
Rendered body type: text/plain

Johan. This message seems just too much to ignore. Have you any
knowledge of this effort and how we might get more information? With
running the HF SIG, is suspect this something others ought to do for the
group. My sucess with BPSK has totally overwhelmed me but I'll try for
more information based on what has already been done. Cliff Buttschardt W6HDO

On Thu, 7 Mar 1996, Sadowski,P O4 USAF JCSAR/EDD wrote:

>
> Saw this particular comment and I couldn't help but throw in my 2 cents.
> US DoD went through extreme pain and agony over the overhead of
> the new protocols over low bandwidth channels. After spending some
> serious money with many different experts... they came up with a low
> overhead protocol... with FEC option... called TACO 2 (tactical comm
> protocol). I suggest the SIG look into this protocol.... DoD has been
> seeking IEEE or ISO standard for this protocol. Its tailor made for
> low BW RF links...
>
> ALOHA
> Al   AH6LS

>
> ----------
> From: hfsig
> To: hfsig
> Subject: [HFSIG:928] Re: Sound Card Interface
> Date: Thursday, February 29, 1996 10:59AM
>
> I see a discussion about ways to implement AX25 for the new HF modems.
> To my opinion thats a bad track. To much overhead, lack of forward error
> correction etc..
> It may seem appealing at the first hand to gain easy access to an existing
> application
> software base, but long time the disadvantages of AX25 will dominate..
> Same is true for TCP/IP... just to much overhead for low thruput channels.
> Makes no sense to put all effort into faster modems and then eat it up
> with protocols that don't fit...
>
> 73 Peter
>
>

34. Pactor/Amtor ARQ

From: "L. J. Shepherd" <ljs@west.net>
Date: Sun, 10 Mar 1996 11:37:50 -0800
To: hfsig@tapr.org
Message-ID: <199603101937.LAA19520@acme.sb.west.net>
Rendered body type: text/plain

Hello Group:
I've been away from the SIG's for a bit and would like to bring myself up-to-date regarding HF DSP-93 software.
I've used Dave Mills excellent AMTOR/SITOR FEC mode B and Rtty modem software with wonderful success. The ability of the DSP-93 to extract what seem to be impossible signals using Dave's sophisticated code is absolutely amazing.
Superlatives aside, I'm currently wondering if AMTOR ARQ or PACTOR has made it to the DSP-93 software collection. In wandering through the ftp sites, the titles do not necessarily describe the program function. There was comment by Johann (some time back) concerning a possible porting of his
PCTOR program to the DSP-93, and I am curious to know if our knowledgable programmers have attempted that task, or perhaps provided any new code for us to experiment with.
This is my first visit to this SIG.......comments will be very much appreciated !
73, Jim K6OYY TAPR 502, AMSAT, ARRL

35. Re: [HFSIG:964] Pactor/Amtor ARQ

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Sun, 10 Mar 1996 16:07:17 -0800
To: hfsig@tapr.org
Message-ID: <9603102358.AA13964@ucs.orst.edu>
Rendered body type: text/plain

Hi Jim,

Welcome to HFSIG.

Frank Perkins has made a modem for the DSP-93 that will work with PCTOR version 3.07. Both are available for downloading from TAPR. This will give you AMTOR ARQ capability.

Just briefly, the PCTOR program is a little different from Dave Mills' approach in that for the TOR program, all you need is to pre-load the DSP modem code with DSPLOAD into the DSP-93 before you fire up the TOR program. The TOR program does the bit and character extraction on the PC side. Dave's program does character extraction on the DSP and uses the DSP-93 UART for communicating character data to a dumb display (PROCOMM or such). Dave's program has some rather unique ideas such as a Viterbi soft-decision decoder for the slicer, which is a pretty sophisticated idea.

There also is another program by Tom Sailer, HB9JNX that does RTTY/AMTOR/Pactor in one program. I have posted Tom's code on the HFSIG program upload area. I'm sure it will work with Frank's HF-93 DSP modem code.

I also am certain that Mike Kerry's, (G4BMK), program(s), BMKMULTY will also work with Frank's DSP-modem because the RS-232 interface that all these TOR programs use are pretty much the same.

I hope that gives you a couple of options to play with. I hope that I can
get to revising and reworking my own HF DSP modems for the DSP-93 in a while.

Let us know how it works out.

73's

--Johan

>Hello Group:
>   I've been away from the SIG's for a bit and would like to
>bring myself up-to-date regarding HF DSP-93 software.
>   I've used Dave Mills excellent AMTOR/SITOR FEC mode B and Rtty
>modem software with wonderful success. The ability of the DSP-93 to extract
>what seem to be impossible signals using Dave's sophisticated code is
>absolutely amazing.
>   Superlatives aside, I'm currently wondering if AMTOR ARQ or
>PACTOR has made it to the DSP-93 software collection. In wandering through the
>ftp sites, the titles do not necessarily describe the program function.
>There was comment by Johann (some time back) concerning a possible porting
>of his
>PCTOR program to the DSP-93, and I am curious to know if our knowledgable
>programmers have attempted that task, or perhaps provided any new code for us
>to experiment with.
>   This is my first visit to this SIG.......comments will be
>very much appreciated !
>   73, Jim   K6OYY   TAPR 502, AMSAT, ARRL
>
>

36. Re: Latest news

From: ajm@teleoscom.com (Anthony J. Monteiro X-6315)
Date: Tue, 12 Mar 96 13:40:00 EST
To: N0AOT@aol.com, forrerj@ucs.orst.edu
Cc: amsat-bb@amsat.org, hfsig@tapr.org
Message-ID: <9603121840.AA18436@teleoscom.com>
Rendered body type: text/plain

I am interested in software for the EVM56002. Do you know of a C compiler
for this that is low cost ? Is there software readily available for
pacsat modems at 1200 or 9600 bps?

Thanks and 73,
Tony AA2TX@amsat.org

37. DSP-93 version HF Channel simulator success

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Tue, 12 Mar 1996 23:09:14 -0800
To: HFSIG@tapr.org
Message-ID: <9603130700.AA09725@ucs.orst.edu>
Rendered body type: text/plain

Hi all,

I thought that I would pass on the good news that the HF simulator code
originally contributed by Alexander, DL8AAU, is now also working very nicely
on the DSP-93.

With the help of Alexander and Juergen, DG1SCR, who made it possble to run
the code the TI 320C25, it still was quite a challenge to get it to run
within the contraints of the 'C25.

When playing a constant carrier into the input, the output really sounds
like a dead carrier on HF with the typical fading and doppler on the signal.

For those that joined HFSIG recently - the HF channel simulator is an
implementation of the Watterson ionospheric model as described in the CCIR
recommendations. It allows several different variations of ionospheric
conditions to be simulated in real time and because the conditions are
repeatable, remains the ideal way to evaluate modem performance under
different conditions.

There remains a bit more work to be done - all I can say is:
Congratulations and good work Alexander and Juergen!

--Johan, KC7WW

38. RE: [HFSIG:967] DSP-93 version HF Channel simulator success

From: "Sadowski,P O4 USAF JCSAR/EDD" <sadowskp@jcsar.nellis.af.mil>
Date: Wed, 13 Mar 96 10:18:00 cst
To: hfsig <hfsig@tapr.org>
Message-ID: <3146F873@mailgtwy.nellis.af.mil>
Rendered body type: text/plain

Question..... At the digital communications conference this topic was discussed for a while... and I seem to recall that there was some discussion as to if this model takes into account noise and inteference in addition to fading and shifts.... At that time, the thought was that in fact no noise or interference was in the CCIR(?) standard.... Lots of mumbling in the audience about the validity of a model that doesn't have different noise and signal interference.... but then something is better than nothing I must admit.... anyway, is there any plan to "improve"
the model with noise and interference???? TNX

Also Johan... anything interesting from the TACO 2 stuff???

ALOHA

Al AH6LS
sadowskp@jcsar.nellis.af.mil

>----------
>From: hfsig
>To: hfsig
>Subject: [HFSIG:967] DSP-93 version HF Channel simulator success
>Date: Wednesday, March 13, 1996 1:13AM

>Hi all,

>I thought that I would pass on the good news that the HF simulator code
>originally contributed by Alexander, DL8AAU, is now also working very nicely
>on the DSP-93.

>With the help of Alexander and Juergen, DG1SCR, who made it possble to run
>the code the TI 320C25, it still was quite a challenge to get it to run
>within the contraints of the 'C25.

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>like a dead carrier on HF with the typical fading and doppler on the signal.

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>implementation of the Watterson ionospheric model as described in the CCIR
>recommendations. It allows several different variations of ionospheric
>conditions to be simulated in real time and because the conditions are
>repeatable, remains the ideal way to evaluate modem performance under
>different conditions.

>There remains a bit more work to be done - all I can say is:
>Congratulations and good work Alexander and Juergen!

>--Johan, KC7WW

39. Re: [HFSIG:937] Re: C5x DSK Assembler

From: dibene@VNET.IBM.COM (Alberto di Bene)
Date: Wed, 13 Mar 96 19:25:57 EST
To: HFSIG@tapr.org
Message-ID: <199603131836.MAA19740@tapr.org>
Rendered body type: text/plain

I contacted Texas Instruments about the bugs in Version 1.10 of DSK5A
 and this is the answer I got. Note that, while the phone BBS may have the
 amended version (I didn't check, too expensive to call from Europe), the ftp
 mirror still has the buggy one, as of today (13th of March).

================================= Cut Here ==================================

 Alberto,

 This is a known bug right after the DSK5A.EXE post on TI modem BBS. A correct
 version of DSK5A.EXE was put on the BBS to replace the wrong one couple days
 later. Please try download the DSK5A.EXE from TI modem BBS to get the right
 version of DSK5A.EXE.

Best Regards,
Jason Chyan

DSP Application Engineer
Texas Instruments Inc.
================================ Cut Here ================================

Alberto di Bene dibene@vnet.ibm.com i2phd@amsat.org

Microsoft gave you windows. OS/2 gives you the whole house.)
---

40. Re: [HFSIG:968] RE: DSP-93 version HF Channel simulator success

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Wed, 13 Mar 1996 13:25:10 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960313130648.31182A-100000@ucs.orst.edu>
Rendered body type: text/plain

Hi Allan,

On Wed, 13 Mar 1996, Sadowski,P O4 USAF JCSAR/EDD wrote:

> Question..... At the digital communications conference this topic was
> discussed for a while... and I seem to recall that there was some
> discussion as to if this model takes into account noise and inteference
> in addition to fading and shifts.... At that time, the thought was that in
> fact no noise or interference was in the CCIR(?) standard.... Lots of
> mumbling in the audience about the validity of a model that doesn't
> have different noise and signal interference.... but then something is
> better than nothing I must admit.... anyway, is there any plan to
> "improve"
> the model with noise and interference???? TNX
>

You make a good point. The CCIR recommendations does not include noise models. I think the discussions at the last DCC on this topic was mostly related to simulating nearby channel interference. This also is not included in the CCIR recommendations.

There are some published work on simulating HF noise using a Markov process in addition to Gaussian noise. The present simulator does include means for computing the incoming signal power and a Gaussian noise generator, so presumeably one can do some Eb/N0 comparisons. The field is wide open and begging for people to work on.

> Also Johan... anything interesting from the TACO 2 stuff??? > > ALOHA >
> Al AH6LS

I spent a bit of time reading the TACO 2 documents - it's volumunous and very involved - the kind of thing that you need someone to walk you through to get the whole picture. I really have not much of an opinion worth anything at this time. There were a couple of first impressions that caught me eye:

1) One must realize that the work was done in 1993, probably then based on older ideas. I think we know now in theory and by practical proof that the ECC methods used for TACO 2 is not the best for the effort, i.e., the block codes it uses.

2) I was surprized to see that it still uses HDLC framing with its weaknesses in its sync vector.

3) It proposes a new networking layering, to replace the exsisting ISO model. This makes it very difficult to judge what the designers were up to when it was proposed.

I do realize that there are good ideas floating around in there, however, we have to get down to asking and answering a couple of fundametal concepts. I have been trying to stimulate some discussion about these issues and things have been coming along slowly. Perhaps this is one item to prepare something for the DCC in September?

Not sure I answered any vexing questions?

--Johan

41. Re: Error Condition Re: unsubscribe

From: Danny Higgins <danny@edstks1.demon.co.uk>
Date: Wed, 13 Mar 1996 15:45:41 +0000
To: hfsig@tapr.org
Message-ID: <TX8fYHAl2uRxEwWY@edstks1.demon.co.uk>
Rendered body type: text/plain

In message <hZV9dBAE9oRxEwRN@edstks1.demon.co.uk>, hfsig@tapr.org writes
>
>We are sorry, but this system sensed the following request which may have been
>inadvertedly sent to this list:
>
>UNSUBSCRIBE
>
>If your posting was intentional, please accept our apologies and resend your
>mail message, making sure you do not include anything that may look like a
>request in the first line of the body of the actual message. If this was
>indeed a request please resend it to listproc@tapr.org
>Your entire message
>is copied below.
>
>--
>Unsubscribe
I wish to unsubscribe for a few weeks.
--
Danny Higgins

42. Re: [HFSIG:970] RE: DSP-93 version HF Channel simulator success

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Thu, 14 Mar 1996 08:52:58 +0100 (MEZ)
To: hfsig@tapr.org
Message-ID: <Pine.A32.3.91.960314084651.20544B-100000@zeus.uet.e-technik.th-darmstadt.de>
Rendered body type: text/plain

On Wed, 13 Mar 1996, Johan Forrer wrote:

>
> Hi Allan,
>
> On Wed, 13 Mar 1996, Sadowski,P O4 USAF JCSAR/EDD wrote:
>
> >
> > Question.....   At the digital communications conference this topic was
> > discussed for a while... and I seem to recall that there was some
> > discussion as to if this model takes into account noise and inteference
> > in addition to fading and shifts....   At that time, the thought was that in
> > fact no noise or interference was in the CCIR(?) standard....   Lots of
> > mumbling in the audience about the validity of a model that doesn't
> > have different noise and signal interference.... but then something is
> > better than nothing I must admit....   anyway, is there any plan to
> > "improve"
> > the model with noise and interference????   TNX
> >
>
> You make a good point. The CCIR recommendations does not include noise
> models. I think the discussions at the last DCC on this topic was mostly
> related to simulating nearby channel interference. This also is not
> included in the CCIR recommendations.
>
> There are some published work on simulating HF noise using a Markov
> process in addition to Gaussian noise. The present simulator does include
> means for computing the incoming signal power and a Gaussian noise
> generator, so presumeably one can do some Eb/N0 comparisons. The field is
> wide open and begging for people to work on.
>

Hi!

Eb/N0 comparision was what I wanted to do with the simulator. Anyway...
It has a very good Gaussian noise generator and enough power to include
some interference models. I have some references about co-channel
interference simulation somewhere. I could i.e. put in some narrow
band Gaussian noise additional to the white noise I add at the
output. The reason why I don't see a need to do so now is that we haven't
got modulation schemes that can handle interference. Ok, we have a
bit of FEC coding but what I think of is an adaptive demodulator that can
withstand a fair amount of narrow band interference.

73' Alexander

43. RE: [HFSIG:968] RE: DSP-93 version HF Channel simulator success

From: Jon Bloom <jbloom@nai.net>
Date: Thu, 14 Mar 1996 04:50:58 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB1162.1973B060@jbloom.nai.net>
Rendered body type: text/plain

Sadowski,P O4 USAF JCSAR/EDD[SMTP:sadowskp@jcsar.nellis.af.mil] wrote:
>
>Question..... At the digital communications conference this topic was
>discussed for a while... and I seem to recall that there was some
>discussion as to if this model takes into account noise and inteference
>in addition to fading and shifts.... At that time, the thought was that in
>fact no noise or interference was in the CCIR(?) standard.... Lots of
>mumbling in the audience about the validity of a model that doesn't
>have different noise and signal interference.... but then something is
>better than nothing I must admit.... anyway, is there any plan to
>"improve"
>the model with noise and interference???? TNX

It's not really a question of the model. What the simulator attempts to do is to model the propagation path. You need this so you can test your system with a desired-signal input that varies like it will when it comes to you by skywave.

The two other things you would like to do is test for the case where you have interfering signals of various types (including non-Gaussian noise sources) and test the system's performance under weak-signal conditions.

Of course, you're likely to want to test using an input to the system that exhibits all of these cases. But adding Gaussian noise and interfering signals is pretty easy from an implementation point of view. That you can do with a couple of op-amps. (Or using DSP, of course.) It is the propagation effects that are hard to do, and that's why I think this report of Johan's is pretty significant.

Testing with interfering "stations" is straightforward. First, select the types of interfering signals you expect to see. In the case of an HF modem for amateur use, this would certainly include signals like those of the system you're testing, but other digital signals (RTTY, AMTOR, packet, etc) and CW signals are likely candidates, too. You also want to consider frequency offsets of the interfering signals from that of the desired signal. Altogether, you can end up with a pretty large test matrix. Concentrating on worst-case scenarios should winnow it down, though. So, you just generate the interfering signal and sum it with the "propagated" desired signal, applying the result to the system under test. (Of course, you could also simulate propagation of the interfering signals, but it seems likely that the worst case would be that of direct, unfaded interfering signals.)

Non-Gaussian (impulsive) noise may be more of a challenge to characterize because it can come from so many different sources, all of which generate noise with different characteristics. But once you've generated your impulse-noise signal, you again just sum it with the test signal.

Finally, weak-signal operation is easy to simulate just by summing bandlimited Gaussian noise with the test signal.

In short, then, the propagation simulator is one key component of the test system, not the entire system.

-- Jon, KE3Z

44. LIST TEST - DON'T READ!

From: Fermax.esp@vlc.servicom.es
Date: Fri, 15 Mar 1996 07:06:01 +0100
To: hfsig@tapr.org
Message-ID: <199603150612.HAA24843@ns.mad.servicom.es>
Rendered body type: text/plain

[Empty message body]

45. Re: [HFSIG:972] RE: HF Channel Interference

From: dona@tridelta.com (Don Adams)
Date: Fri, 15 Mar 1996 08:43:47 -0500 (EST)
To: hfsig@tapr.org
Message-ID: <199603151343.IAA11659@sparcy.tridelta.com>
Rendered body type: text/plain

I think we should worry about the performance over basicaly a clean HF path
first. ie low gaussian s/n. Once we get something that has the performance
we want ie high bit rates on good channels and low but get thru bit rates
on weak channels, we can then use some sort of spread spectrum to spread
the interfering signal. The spreading sequence might be optimized, (or may not

have to be optimized), for the particular interference. I think we could
handle this
as a separate issue.

>
>Eb/N0 comparision was what I wanted to do with the simulator. Anyway...
>It has a very good Gaussian noise generator and enough power to include
>some interference models. I have some references about co-channel
>interference simulation somewhere. I could i.e. put in some narrow
>band Gaussian noise additional to the white noise I add at the
>output. The reason why I don't see a need to do so now is that we haven't
>got modulation schemes that can handle interference. Ok, we have a
>bit of FEC coding but what I think of is an adaptive demodulator that can
>withstand a fair amount of narrow band interference.
>
>73' Alexander
>
>
>

46. HF Simulator Examples uploaded

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Sat, 16 Mar 1996 09:36:14 -0800
To: HFSIG@tapr.org
Message-ID: <9603161727.AA21829@ucs.orst.edu>
Rendered body type: text/plain

Hi,

I have made a couple of examples of HF channel simulation for use on the
DSP-93. It consists of flat and selective fading conditions of good,

moderate, and poor severity. These are on the HFSIG upload area as
"SIMTEST.TXT" and "SIMTEST.ZIP"

These are loadable objects that you load with the DSP-93's "DSPLOAD"
program. If you have a DSP-93, please try it - we will apprecaite some feedback.

73's and good luck.

--Johan, KC7WW

47. DSP-93 HF Simulators

From: "Mark L. Hammond" <hammond@camel.campbell.edu>
Date: Sun, 17 Mar 1996 02:03:27 +0000
To: dsp-93@tapr.org, hfsig@tapr.org
Message-ID: <1.5.4b12.32.19960317020327.0067806c@camel.campbell.edu>
Rendered body type: text/plain

Hi Johan,

I tried the three simulations on my DSP-93. Sounds pretty nifty. I
listened to each of them on a variety of audio signals on my IC-706--2M
repeater(!), FM white noise, HF SSB white noise, hf digital signals, etc.

Not sure what kind of feedback you want, but the simulators do a great
representation to my ears, at least, of sounding like a variety of HF band
conditions--fades, dips, multipath, etc.

(btw, I'm not on the hfsig mailing group--only the dsp-93 group).

>Hi,
>
>I have made a couple of examples of HF channel simulation for use on the
>DSP-93. It consists of flat and selective fading conditions of good,
>moderate, and poor severity. These are on the HFSIG upload area as
>"SIMTEST.TXT" and "SIMTEST.ZIP"
>
>These are loadable objects that you load with the DSP-93's "DSPLOAD"
>program. If you have a DSP-93, please try it - we will apprecaite some
feedback.
>
>73's and good luck.
>
>--Johan, KC7WW
>
>
>
>

Mark L. Hammond [KC4EBR]

48. Re: [HFSIG:977] DSP-93 HF Simulators

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Sun, 17 Mar 1996 21:10:19 -0800
To: hfsig@tapr.org
Message-ID: <199603180500.VAA27509@PEAK.ORG>
Rendered body type: text/plain

Hi Mark,

Thanks for the feedback - that's a good start.

>Hi Johan,
>
>I tried the three simulations on my DSP-93. Sounds pretty nifty. I
>listened to each of them on a variety of audio signals on my IC-706--2M
>repeater(!), FM white noise, HF SSB white noise, hf digital signals, etc.
>
>Not sure what kind of feedback you want, but the simulators do a great
>representation to my ears, at least, of sounding like a variety of HF band
>conditions--fades, dips, multipath, etc.
>
>(btw, I'm not on the hfsig mailing group--only the dsp-93 group).
>

Do you think the audio gain settings is OK?

--Johan

49. subscribe

From: Reinier van der Lee <rvdlee@austria.cp.philips.com>
Date: Tue, 19 Mar 1996 09:46:11 +-100
To: "'ham-tech@netcom.com'" <ham-tech@netcom.com>, "'TAPR'" <hfsig@tapr.org>,
        "'boston radio club'" <listserv@netcom.com>
Message-ID: <01BB1578.F706D6E0@Reinier>
Rendered body type: text/plain

subscribe


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M`P`V``````! `#D`H"HZAG 5NP$>`' ``0````H```!S=6)S8W)I8F4````"
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MSQ[_(L0Y,B8%#R=Q'Y,G< *"<W1Y;#4'D&@)X'0```/P9&,\=&P*LP^#`= #
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M8B H`I%ACU.#-4#_7S]C_V4/9A]3L#P09V)4/_]HSVG?5KP?<&=O;.]M_V\/
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$``!F( ``
`
end
```

50. Re: [HFSIG:978] Re: DSP-93 HF Simulators

From: Per-Tore <c-three@telepost.no>
Date: Thu, 21 Mar 1996 13:27:16 +0100
To: hfsig@tapr.org
Message-ID: <199603211227.26397.telepost.no@telepost.no>
Rendered body type: text/plain

I am new to this list, and would like to have more information on the DSP-93
system. Where can I get it, what does it cost, what software is already
available?

Regards,
Per-Tore
LA7NO


```
-----------------------------------+--------------------------------
| Per-Tore Aasestrand              | E-mail :  c-three@telepost.no |
| C-Three Systems AS               | Fax    :  + 47 - 55 23 18 88  |
| P.O. Box 1708                    | Voice  :  + 47 - 55 32 67 67  |
| N-5024 Bergen, Norway            | Mobile :  + 47 - 92 04 44 76  |
-----------------------------------+--------------------------------
```


51. Help - HF TNC

From: dblunn@opennet.net.au (Dave Blunn)
Date: Fri, 22 Mar 1996 19:51:33 +1000
To: hfsig@tapr.org
Message-ID: <199603220951.TAA07776@Tandem1.opennet.net.au>
Rendered body type: text/plain

Can anyone advise my on which is the better of the two HF TNCs, as I am
new to HF digital OPS.

A PK 232 MBX or a KAM Plus ?

The PK 232 is the cheaper of the two but does not have G-TOR, is it
worth the extra to have G-TOR ?

Regards

Dave Blunn
VK2DDJ

52. Re: [HFSIG:981] Help - HF TNC

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Fri, 22 Mar 1996 09:18:11 +0600 (GMT)
To: hfsig@tapr.org
Message-ID: <199603220318.JAA21086@www.kelly-afb.org>
Rendered body type: text/plain

David,

In your message you write:
>
>
> Can anyone advise my on which is the better of the two HF TNCs, as I am
> new to HF digital OPS.
>
> A PK 232 MBX or a KAM Plus ?

The PK-232 is build better, physically, but that doesn't mean it
will last longer (MTBF).

>
> The PK 232 is the cheaper of the two but does not have G-TOR, is it
> worth the extra to have G-TOR ?

G-TOR is far superior to packet and might be better than Pactor
or Pactor II.

Don't try using packet at 300 baud on HF, this flies in the face
of research done back in the late 40's and early 50's concerning
baud rates on HF. SRI, Collins and other discovered that HF
propagation, the ionspheric movement, distorts HF signals and
causes multi-path signals that can cause FSK/AFSK signals to be
un-detectable at baud rates above 100 and surely 150 baud. Of
course, under good conditions, you would have the problem. I've
heard 20 over S9 signals on HF and no QRM and the packet signal
at 300 baud wouldn't decode. Dropped the buad rate to 110 baud
and 100% copy.

BTW, both Pactor and G-TOR are "smart" modes as they automatically
adjust baud rates if signals become un-decodable.

73, Walt/K5YFW

>
>
>
> Regards
>
> Dave Blunn
> VK2DDJ
>
>

53. RE: [HFSIG:981] Help - HF TNC

From: Stephan Sykes <ssykes@ns2.emirates.net.ae>
Date: Sat, 23 Mar 1996 06:57:12 +-400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB1885.F9FC5040@csb025.emirates.net.ae>
Rendered body type: text/plain

In tests I was involved in a couple of years ago, the modem in the AEA was somewhat better for long haul
 packet. It probably comes down to personal preference though, each unit has unique features.

From: Dave Blunn[SMTP:dblunn@opennet.net.au]
Sent: Friday, March 22, 1996 8:04 AM
To: hfsig@tapr.org
Subject: [HFSIG:981] Help - HF TNC

Can anyone advise my on which is the better of the two HF TNCs, as I am
new to HF digital OPS.

A PK 232 MBX or a KAM Plus ?

The PK 232 is the cheaper of the two but does not have G-TOR, is it
worth the extra to have G-TOR ?

Regards

Dave Blunn
VK2DDJ

54. Updated HF Channel Simulator

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Sat, 23 Mar 1996 08:53:08 -0800
To: HFSIG@tapr.org
Message-ID: <9603231642.AA07325@ucs.orst.edu>
Rendered body type: text/plain

Hi,

For those interested in the DSP-93 version of the HF channel simulator,
please check out the latest upload of an experimental HF channel simulator

test.

With Alexander's (DL8AAU) help, I managed to fix a few errors, and also
included some new test conditions. The examples now reflect the CCIR
specifications more accurately and now include:

Selective fading: Good, moderate, poor, flutter fading,
two kinds of flat fading: Good and severe.

The updated package is available for downloading from:
ftp.tapr.org/tapr/SIG/HFSIG/upload/simtest1.zip

Trust you find it useful.

73's

--Johan Forrer, KC7WW

55. Re: [HFSIG:984] Updated HF Channel Simulator

From: k6sti@n2.net (Brian Beezley)
Date: Sat, 23 Mar 1996 11:26:53 -0800
To: hfsig@tapr.org
Message-ID: <199603231926.LAA20864@ravel.n2.net>
Rendered body type: text/plain

>Hi,
>
>For those interested in the DSP-93 version of the HF channel simulator,
>please check out the latest upload of an experimental HF channel simulator
>test.
>
>With Alexander's (DL8AAU) help, I managed to fix a few errors, and also
>included some new test conditions. The examples now reflect the CCIR
>specifications more accurately and now include:
>
>Selective fading: Good, moderate, poor, flutter fading,
>two kinds of flat fading: Good and severe.
>
>The updated package is available for downloading from:
>ftp.tapr.org/tapr/SIG/HFSIG/upload/simtest1.zip
>
>Trust you find it useful.
>
>73's
>
>--Johan Forrer, KC7WW
>
>
>

Johan, HFSIG needs to be hfsig in the URL. Thanks for the file.

                    73--Brian, K6STI
                        k6sti@n2.net

56. Announcement

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Sun, 24 Mar 1996 12:19:50 -0800
To: HFSIG@tapr.org
Message-ID: <9603242009.AA27931@ucs.orst.edu>
Rendered body type: text/plain

Hi all,

After talking with Greg - an update that may or may not be of interest to
you; I will be finishing my work in academia around June (1996) and then be
pursuing opportunities with my own company in future.

The reason that this concerns HFSIG is that I intend doing future
development work in the field of HF digital communications on a professional
basis. It is quite likely that a conflict of interest could develop on
certain issues, particularly as this special interest group is monitored by
several with similar commercial interests.

So - where does that leave HFSIG? I am willing to act as moderator for a
while at least, however, invite your ideas and suggestions. If you would
like to nominate someone that would be willing to take the reigns, please

contact either me of Greg.

Your consideration is much appreciated.

--Johan Forrer, KC7WW

----00----

P.S. In case you wonder; I will be spending a substantial amount of time on "new" HF digital communication systems for a U.S-based client, act as a DSP and hardware/software consultant, and be actively developing a family micromachine-agents on my own. If you perhaps know of anyone that needs help, please spread the word and keep me in mind in future.

57. New EVM software package EVM56K3

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 25 Mar 1996 09:07:19 -0800
To: HFSIG@tapr.org
Message-ID: <63F82434FCD@frl.orst.edu>
Rendered body type: text/plain

Hi all,

I have had so many requests for various items for use with the DSP56002EVM that I made an updated version and uploaded it to HFSIG upload area.

The new software includes a new operating system (Leonid) that fixes some bugs, also allows for users to select the 66 and 80 MHz DSP chips. There is an expanded interface schematic and updated modem codes.

Be sure to check out the new weak signal detection and display software from Pawel, SP9VRC - it is superb!

The code package is on: ftp.tapr.org/tapr/SIG/hfsig/upload/evm56k3.zip

Enjoy.

--Johan, KC7WW

58. Re: [HFSIG:987] New EVM software package EVM56K3

From: dwborden@mail.pananet.com (David Borden (58))
Date: Tue, 26 Mar 1996 09:37:48 -0500
To: hfsig@tapr.org
Message-ID: <3158013C.4D6E@pananet.com>
Rendered body type: text/plain

Johan,
  How do I print this evm56k post script schematic and pcb stuff?
It does not appear to be Post Script, it looks binary encoded (after I run pkunzip on it of course). I think I got it from evm56k2.ps on the TAPR ftp directory. I tried sending it to my Post Script printer, but it declared it bad........david,k8mmo/hp2

Johan Forrer wrote:
>
> Hi all,
>
> I have had so many requests for various items for use with the DSP56002EVM
> that I made an updated version and uploaded it to HFSIG upload area.
>
> The new software includes a new operating system (Leonid) that fixes
> some bugs, also allows for users to select the 66 and 80 MHz DSP chips.
> There is an expanded interface schematic and updated modem codes.
>
> Be sure to check out the new weak signal detection and display software from
> Pawel, SP9VRC - it is superb!
>
> The code package is on: ftp.tapr.org/tapr/SIG/hfsig/upload/evm56k3.zip
>
> Enjoy.
>
> --Johan, KC7WW

59. HF RDF

From: Sean Warren-Cox (SWC) <seanw@hpqs0212.sqf.hp.com>
Date: Tue, 26 Mar 1996 17:05:51 GMT
To: hfsig@tapr.org
Message-ID: <199603261705.AA116909952@hpqs0212.sqf.hp.com>
Rendered body type: text/plain

Hi all,

I may get laughed of the stage for this but does anybody
know of a way of getting position information of a
transmitter to sub 1cm accuracy. I want to measure distances through
solid objects such as walls in a house. My thoughts were to
use Radio Direction Finding but I guess that for the accuracy I require
the frequencies involved would be in the 5 GHz range and not very good for
transmitting though solid objects. Are there any tricks used to
calculate distance and position from a transmitted signal at lower
frequencies.

Thanks for your time,

Sean
____

          Sean Warren-Cox | sean@hpiplaa.sqf.hp.com
          TSD direct line | tel: 0131 331 7080 (Edinburgh, Scotland)
____

60. Re: [HFSIG:989] HF RDF

From: Per-Tore <c-three@telepost.no>
Date: Wed, 27 Mar 1996 11:21:50 +0100
To: hfsig@tapr.org
Message-ID: <199603271021.16028.telepost.no@telepost.no>
Rendered body type: text/plain

At 11:15 26.03.96 -0600, you wrote:
>Hi all,
>
>I may get laughed of the stage for this but does anybody
>know of a way of getting position information of a
>transmitter to sub 1cm accuracy. I want to measure distances through
>solid objects such as walls in a house. My thoughts were to
>use Radio Direction Finding but I guess that for the accuracy I require
>the frequencies involved would be in the 5 GHz range and not very good for
>transmitting though solid objects. Are there any tricks used to
>calculate distance and position from a transmitted signal at lower
>frequencies.
>
>Thanks for your time,
>
>Sean
>____
>
>   Sean Warren-Cox | sean@hpiplaa.sqf.hp.com
>   TSD direct line | tel: 0131 331 7080 (Edinburgh, Scotland)
>____

Hi Sean,

No, this is not a rediculous idea at all!
In my opinion there are two ways to go in order to achieve the necessary
accuracy:

1) As you suggested, use VERY high frequency.

2) Use Direct-Sequence-Spread-Spectrum with a sufficiently high chip-rate.

In fact, using DSSS one can achieve higher precision than with any other
method - with the exeption of using a laser, of course!

Hope this helps.

Regards,

Per-Tore
LA7NO

----------------------------------+--------------------------------
| Per-Tore Aasestrand | E-mail : c-three@telepost.no |
| C-Three Systems AS | Fax   : + 47 - 55 23 18 88 |

```
| P.O. Box 1708                   | Voice  :  + 47 - 55 32 67 67  |
| N-5024 Bergen, Norway           | Mobile :  + 47 - 92 04 44 76  |
----------------------------------+--------------------------------
```

61. Sorry guys

```
From: Mike Reublin <nf4l@jaxnet.com>
Date: Wed, 27 Mar 1996 17:27:11 -0500
To: hfsig@tapr.org
Message-ID: <3159C0BF.5717@jaxnet.com>
Rendered body type: text/plain
```

```
I really didn't want to do this, and stand here in my Nomex awaiting the
flames, but I can't think of any alternative.

I've been trying for many weeks to unsubscribe from this group. Message comes
back from the list server:

NF4L@JAXNET.COM: You are not subscribed to hfsig@tapr.org

In addition, the system found the following address(es) that resemble yours.
If one of these is you, please resend your message from that one, or use the
'set <list> address' request to change the address you are subscribed with:

NF4L@MAILHUB.JAXNET.COM

I can get mail at either address (they're the same re: incoming). I reset my
mailer to the mailhub one, sent the unsubscribe again. Same result. Sent for
the help file. No help there. Tried the "set address" a couple of times,
errrors returned. Sent 2 messages to "if you're having trouble". No answer.

Could one of you gentlemen (no pileups) give me a way out?

Thanks, and again my apologies for taking up the BW.

                        73 de Mike NF4l
--
*************************************************************************
Mike Reublin                                        email: nf4l@jaxnet.com
Senior Programmer/Analyst                     Alltel Information Services
*************************************************************************
```

62. Coherent CW

```
From: deh6@po.CWRU.Edu (Donald E. Haselwood)
Date: Fri, 29 Mar 1996 11:06:20 -0500
To: HFSIG@tapr.org
Message-ID: <199603291606.LAA06593@kanga.INS.CWRU.Edu>
Rendered body type: text/plain
```

```
I'm new on this sig, but I've been on the DSP-93 sig for "ages,"
have a DSP93 going (but more an educational toy than part of the
station setup), and I've been tinkering/looking up references
having to do with cw decoding.  Coherent cw came to my attention.

Can anyone direct me to info on coherent cw (or just digging cw
out of the noise such as EME, HF, etc.)?

I suspect there are "batches" of messages someplace in the HFSIG
archives, but wading through them would be a project in itself.

73's
Don, K4JPJ
```

63. (no subject)

```
From: (unknown sender)
Date: (date unavailable)
Message-ID: <199603291855.KAA02584@mh1.lbl.gov>
Rendered body type: text/plain
```

```
Date: Fri, 29 Mar 1996 10:54:52 -0800
To: hfsig@tapr.org
From: Mike Fahmie <MPFahmie@lbl.gov>
Subject: Re: [HFSIG:992] Coherent CW

At 10:15 AM 3/29/96 -0600, Don (K4JPJ) wrote:
:
:
>Can anyone direct me to info on coherent cw (or just digging cw
>out of the noise such as EME, HF, etc.)?
```

>

If you have a good library (or you happen to have them laying around the
shack), you can read these articles to get the basic idea.

QST   Sept '75   page 26 Coherent CW, Amateur Radio's New State of
the Art?
CQ   June/July'77   Coherent CW - The CW of the Future
QST   May '81   page 11 Coherent CW - The Concept (part 1)
QST   June'81   page 18 Coherent CW - The Practical Aspects
QEX   Feb'94   A DSP Version of Coherent CW

-Mike-

Mike Fahmie   -   Advanced Light Source
Lawrence Berkeley National Laboratory
(510) 486-4030   -   MPFahmie@lbl.gov
WA6ZTY @ N6EEG

64. Re: [HFSIG:986] Announcement

From: Clifford Buttschardt <cbuttsch@slonet.org>
Date: Fri, 29 Mar 1996 15:49:31 -0800 (PST)
To: Johan Forrer <forrerj@ucs.orst.edu>
Cc: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960329154443.28325A-100000@biggulp.callamer.com>
Rendered body type: text/plain

Johan, as you I will be done with academia in June. I've been doing
some, but not a great deal of consulting work. I would like to talk
further with you regarding some common effort, especially so since I will
be moving North of you just a few dozen miles. The server has been down
since the 24th, so best I call you. If we miss then, Dr. Cliff
Buttschardt, Cal Poly, SLO, CA Engineering Technology (805) 772 2132.

On Sun, 24 Mar 1996, Johan Forrer wrote:

> Hi all,
>
> After talking with Greg - an update that may or may not be of interest to
> you; I will be finishing my work in academia around June (1996) and then be
> pursuing opportunities with my own company in future.
>
> The reason that this concerns HFSIG is that I intend doing future
> development work in the field of HF digital communications on a professional
> basis. It is quite likely that a conflict of interest could develop on
> certain issues, particularly as this special interest group is monitored by
> several with similar commercial interests.
>
> So - where does that leave HFSIG? I am willing to act as moderator for a
> while at least, however, invite your ideas and suggestions. If you would
> like to nominate someone that would be willing to take the reigns, please
> contact either me of Greg.
>
>
> Your consideration is much appreciated.
>
> --Johan Forrer, KC7WW
>
>   ----00----
> P.S. In case you wonder; I will be spending a substantial amount of time on
> "new" HF digital communication systems for a U.S-based client, act as a DSP
> and hardware/software consultant, and be actively developing a family
> micromachine-agents on my own. If you perhaps know of anyone that needs
> help, please spread the word and keep me in mind in future.
>
>
>
>

65. Re: [HFSIG:989] HF RDF

From: Clifford Buttschardt <cbuttsch@slonet.org>
Date: Fri, 29 Mar 1996 17:14:48 -0800 (PST)
To: Sean Warren-Cox <seanw@hpqs0212.sqf.hp.com>
Cc: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960329171005.28325I-100000@biggulp.callamer.com>
Rendered body type: text/plain

Sean, there was an absolutly superb experiment run by Stanford Research
Institute when I was still with them in which solid objects were to be
triangulated within some of the pyramids in Egypt! Various frequencies were
used as well. By my simply mentioning this on the net, I would suppose
we will get an answer, but if not, I'll see if I can get a clue on this
as time passes. Cliff Buttschardt W6HDO

On Tue, 26 Mar 1996, Sean Warren-Cox wrote:

> Hi all,
>
> I may get laughed of the stage for this but does anybody
> know of a way of getting position information of a
> transmitter to sub 1cm accuracy. I want to measure distances through
> solid objects such as walls in a house. My thoughts were to
> use Radio Direction Finding but I guess that for the accuracy I require
> the frequencies involved would be in the 5 GHz range and not very good for
> transmitting though solid objects. Are there any tricks used to
> calculate distance and position from a transmitted signal at lower
> frequencies.
>
> Thanks for your time,
>
> Sean
> ____
>
>   Sean Warren-Cox | sean@hpiplaa.sqf.hp.com
>   TSD direct line | tel: 0131 331 7080 (Edinburgh, Scotland)
> ____
>
>

66. Re: [HFSIG:992] Coherent CW

From: Clifford Buttschardt <cbuttsch@slonet.org>
Date: Sat, 30 Mar 1996 16:31:01 -0800 (PST)
To: "Donald E. Haselwood" <deh6@po.cwru.edu>
Cc: Bill DeCarle <bill@ietc.ca>, hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960330161757.24987A-100000@biggulp.callamer.com>
Rendered body type: text/plain

Donald: I am sorry to say that nearly nothing has been posted on the
HFSIG and the DSP93 SIG regarding Coherent CW. It simply has been too
awkward to do so with the further observation that DSP93 programming has
been less than startling to say the least. Personally, I would like to
see the DSP93 engine involved in this effort as would many others.
     In the meantime, consult QEX for Feb 1994 for a start in this
effort. Also read what has been done years ago by DeCarle in the Jan
1992 issue of QST. You will appreciate the fact that we have been doing
this very type of communicating for years without the hassel of a devoted
chip and using the power of a PC instead. You will find experiments in
the 1750 meter band using BPSK (DPSK) and on all the ham bands using
Coherent CW observing the 35 KHz and up rule. You might want to consult
the Digital Journal for a monthly column addressed to these items as well.
You will find a great deal of accomplishment has occured with very little
talk just for the sake of talking! 73 Cliff Buttschardt W6HDO

On Fri, 29 Mar 1996, Donald E. Haselwood wrote:

> I'm new on this sig, but I've been on the DSP-93 sig for "ages,"
> have a DSP93 going (but more an educational toy than part of the
> station setup), and I've been tinkering/looking up references
> having to do with cw decoding. Coherent cw came to my attention.
>
> Can anyone direct me to info on coherent cw (or just digging cw
> out of the noise such as EME, HF, etc.)?
>
> I suspect there are "batches" of messages someplace in the HFSIG
> archives, but wading through them would be a project in itself.
>
> 73's
> Don, K4JPJ
>
>

67. Need simple PTT circuit

From: Mike Hooper <mhooper@netcom.com>
Date: Sat, 30 Mar 1996 17:39:56 -0800
To: hfsig@tapr.org

Message-ID: <1.5.4b12.32.19960331013956.0068d0d8@netcom.com>
Rendered body type: text/plain

Hello Everyone:

I am experimenting with Tom Sailor's PSATNC driver. The driver does not provide for PTT keying from a COMPORT as does PSATOR. While PTT data can be obtained from the FLAG_OUT pin on the DSP 2115 chip I would rather use a simple circuit that utilizes the tones going to the radio, rectifies them, and drives a simple 2n2222a switch that takes the radio's PTT to ground.

Your assistance in this matter would be greatly appreciated.

-73- Mike

Mike Hooper
mhooper@netcom.com
kf6pu@wb6ymh.#soca.ca.usa.noam

68. Re: [HFSIG:996] Re: Coherent CW

From: deh6@po.CWRU.Edu (Donald E. Haselwood)
Date: Sat, 30 Mar 1996 21:21:17 -0500
To: hfsig@tapr.org
Message-ID: <199603310221.VAA10080@kanga.INS.CWRU.Edu>
Rendered body type: text/plain

Cliff,

Thanks for sending the message.

>Also read what has been done years ago by DeCarle in the Jan
>1992 issue of QST. You will appreciate the fact that we have
>been doing this very type of communicating for years without the
>hassel of a devoted chip and using the power of a PC instead.

I pulled Jan '92 QST then it "clicked." I ordered the PC board for this project, but never assembled it (as I remember there were a couple of parts missing the catalogs I had, plus I was in the first two years of a Phd program at Case Western Reserve U. and pressed for time. It was an intriguing project but it fell by the wayside. [I'm retired and working on PhD in Organizatonal Behavior as "fun project." I have a MSEE and did engineering with data stuff and computers, starting in 1960 (before you could plug into the switched network, and hardly anybody had heard the term modem, etc.). I moved in management, got an early retirement, etc.] I'm past the "qualifying" stage on the degree so time is my own, though I still have to keep at it to get the dissertation done.

Someone else posted some QST, QEX, and CQ references. I have the QEX article and that outlines the basics. I remember, way back, reading about a telegrapher "trick" called "doubling on the drop" or some such term, where the experienced telegraphers could double the speed of a telegraph circuit by operating synchronously. It was probably a forerunner of coherent cw.

>You will find experiments in the 1750 meter band using BPSK
>(DPSK) and on all the ham bands using Coherent CW observing the
>35 KHz and up rule. You might want to consult the Digital
>Journal for a monthly column addressed to these items as well.

Digital Journal is new to me. I'll keep my eye open for it.

73's
Don

69. Improved DSK (320C26) Assembler

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Sat, 30 Mar 1996 20:48:16 -0800
To: HFSIG@tapr.org
Message-ID: <199603310436.UAA26687@PEAK.ORG>
Rendered body type: text/plain

Hi,

For those working with the TI 320C26 DSK, here is an alternative and improved assembler to the limited one supplied with the DSK.

Thanks to Mike, G4BMK, for contributing his masterpiece.

Look for Mike's assembler at:

ftp.tapr.org/tapr/SIG/hfsig/upload/maxas.zip

Enjoy,

--Johan

70. Skeds with EVM/DSP4

From: Sivula Timo <sivula@NCSNRS01ES.NTC.nokia.com>
Date: Sun, 31 Mar 96 14:20:00 eet
To: "'hfsig'" <hfsig@tapr.org>
Message-ID: <315E69B6@ntcite01es.ntc.nokia.com>
Rendered body type: text/plain

Hello,

is there anybody out there who would be interested in having a sked to
Finland using some of Pawels, SP9VRC?s software on a EVM or DSP4? Paulus,
PA0OCD and I have had a lot of QSO?s and it would be nice to have some more
people in the skeds as well.

regards

Timo, OH2LVZ

Timo.Sivula@ntc.nokia.com

71. Re: [HFSIG:997] Need simple PTT circuit

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Sun, 31 Mar 1996 08:50:42 -0800
To: hfsig@tapr.org
Message-ID: <9603311639.AA30111@ucs.orst.edu>
Rendered body type: text/plain

Hello Mike,

>From what I understood from Tom (and it was some long time ago), was that
the time delay involved for the PTT command to work it's way from the DSP
software, to the PC, to the PTT switch on the COM port, was unsatisfactory.
That is because the sound card's DSP is doing HDLC and time slot persistance
control and does not originate from the PC. Thus the delay involved and why
he choose to add the PTT switch via the flag-out pin from where the DSP can
directly control PTT. I'm not sure whether TOM's program stilll works like
that.

You may want to e-mail Tom and ask him for a solution. I agree that the COM
port-derived PTT is the simplest way to go.

Good luck, 73's

--Johan

>Hello Everyone:
>
> I am experimenting with Tom Sailor's PSATNC driver. The driver does not
>provide for PTT keying from a COMPORT as does PSATOR. While PTT data can be
>obtained from the FLAG_OUT pin on the DSP 2115 chip I would rather use a simple
>circuit that utilizes the tones going to the radio, rectifies them, and drives
>a simple 2n2222a switch that takes the radio's PTT to ground.
>
> Your assistance in this matter would be greatly appreciated.
>
>-73- Mike
>
>
>Mike Hooper
>mhooper@netcom.com
>kf6pu@wb6ymh.#soca.ca.usa.noam
>
>

72. Re: [HFSIG:998] Re: Coherent CW

From: Clifford Buttschardt <cbuttsch@slonet.org>
Date: Sun, 31 Mar 1996 08:49:31 -0800 (PST)
To: "Donald E. Haselwood" <deh6@po.cwru.edu>

Cc: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960331084304.31210A-100000@biggulp.callamer.com>
Rendered body type: text/plain

Hello Don. Yes we are making very good progress with simple effort using
both BPSK and CCW (BiPhase and Coherent CW). Since you started the
process some years back why not now complete the proceedure? Regarding
the DeCarle Spectrum analyzer board, the difficult chip has been the
RS232 driver. Bill in Canada has a supply of them and I see where most
mail order houses now carry the MAX equivelent chip which still fits the
boards you purchase four years ago!

Your next step is to check the home page of VE2IQ:
     http://w3ietc.ca/home/bill/spbill.htm

All the software you need and full instructions for what we are doing is
easily downloaded without hassel.

Good luck on ya and do join us "ordinary" guys! Cliff Buttschardt W6HDO

73. Re: [HFSIG:1002] Re: Coherent CW

From: forrerj@ucs.orst.edu (Johan Forrer)
Date: Sun, 31 Mar 1996 17:30:39 -0800
To: hfsig@tapr.org
Message-ID: <9604010118.AA12405@ucs.orst.edu>
Rendered body type: text/plain

Hi Cliff,

>Hello Don. Yes we are making very good progress with simple effort using
>both BPSK and CCW (BiPhase and Coherent CW). Since you started the
>process some years back why not now complete the proceedure? Regarding
>the DeCarle Spectrum analyzer board, the difficult chip has been the
>RS232 driver. Bill in Canada has a supply of them and I see where most
>mail order houses now carry the MAX equivelent chip which still fits the
>boards you purchase four years ago!
>

Could you verify this URL please: I don't seem to be able to locate the host
"w3ietc.ca"

>Your next step is to check the home page of VE2IQ:
>   http://w3ietc.ca/home/bill/spbill.htm
>

Thanks,

--Johan

74. Re: [HFSIG:1003] Re: Coherent CW

From: Clifford Buttschardt <cbuttsch@slonet.org>
Date: Sun, 31 Mar 1996 18:23:10 -0800 (PST)
To: Johan Forrer <forrerj@ucs.orst.edu>
Cc: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960331181731.13065H-100000@biggulp.callamer.com>
Rendered body type: text/plain

Hi Johan. Good to see you in print. I hope I have made a mistake leaving
out a decimal point as that would answer you request. Here is his URL:
       http://w3.ietc.ca/home/bill/spbill.htm
John, I used ordinary LYNX earlier today and was sucesful getting to
him. I'll check my outgoing mail and see if I left off the decimal point.
Do try and see if some of this information is useful to the group. It
may not be all that can be done but it certainly is a useful start!
Cliff Buttschardt   W6HDO

On Sun, 31 Mar 1996, Johan Forrer wrote:

> Hi Cliff,
>
> >Hello Don. Yes we are making very good progress with simple effort using
> >both BPSK and CCW (BiPhase and Coherent CW). Since you started the
> >process some years back why not now complete the proceedure? Regarding
> >the DeCarle Spectrum analyzer board, the difficult chip has been the
> >RS232 driver. Bill in Canada has a supply of them and I see where most
> >mail order houses now carry the MAX equivelent chip which still fits the

> >boards you purchase four years ago!
> >
>
> Could you verify this URL please: I don't seem to be able to locate the host
> "w3ietc.ca"
>
>
>
> >Your next step is to check the home page of VE2IQ:
> >   http://w3ietc.ca/home/bill/spbill.htm
> >
>
>
> Thanks,
>
> --Johan
>
>

75. Re: BPSK, Re: Coherent CW

From: Clifford Buttschardt <cbuttsch@slonet.org>
Date: Sun, 31 Mar 1996 18:54:30 -0800 (PST)
To: "Donald E. Haselwood" <deh6@po.cwru.edu>,
        Johan Forrer <forrerj@ucs.orst.edu>, hfsig@tapr.org
Cc: Bill DeCarle <bill@ietc.ca>
Message-ID: <Pine.OSF.3.91.960331184516.10946A-100000@biggulp.callamer.com>
Rendered body type: text/plain

Don and Johan, et al:
In addition to Bill DeCarles home page at
http://w3.ietc.ca/home/bill/spbill.htm
you might want to see the Longwave Club of America BBS at:
http://users.aol.com/lwcanews/lwcanews.html

You will find the information here rather informative especially for
those interested in experimentation outside of the ham frequencies. Both
sites have been confirmed just today as valid. Cliff Buttschardt W6HDO

April 1996

Original source file: hfsig.9604
Source: https://archive.org/details/tapr-1997
Recovered from: https://archive.org/download/TAPR-HF_1996-04/TAPR-HF_1996-04.txt

1. Re: [HFSIG:1002] Re: Coherent CW

From: Tom Watson <twatson@magic1.org>
Date: Mon, 01 Apr 1996 10:21:40 -0600
To: hfsig@tapr.org
Message-ID: <31600294.50D9@magic1.org>
Rendered body type: text/plain

The link to VE2IQ does not have a DNS entry!

Tom, WA0DJP

Clifford Buttschardt wrote:
>
> Hello Don. Yes we are making very good progress with simple effort using
> both BPSK and CCW (BiPhase and Coherent CW). Since you started the
> process some years back why not now complete the proceedure? Regarding
> the DeCarle Spectrum analyzer board, the difficult chip has been the
> RS232 driver. Bill in Canada has a supply of them and I see where most
> mail order houses now carry the MAX equivelent chip which still fits the
> boards you purchase four years ago!
>
> Your next step is to check the home page of VE2IQ:
>   http://w3ietc.ca/home/bill/spbill.htm
>
> All the software you need and full instructions for what we are doing is
> easily downloaded without hassel.
>
> Good luck on ya and do join us "ordinary" guys! Cliff Buttschardt W6HDO

2. Re: [HFSIG:1003] Re: Coherent CW

From: Tom Watson <twatson@magic1.org>
Date: Mon, 01 Apr 1996 10:27:45 -0600
To: hfsig@tapr.org
Message-ID: <31600401.2F1B@magic1.org>
Rendered body type: text/plain

Johan Forrer wrote:
>
> Hi Cliff,
>
> >Hello Don. Yes we are making very good progress with simple effort using
> >both BPSK and CCW (BiPhase and Coherent CW). Since you started the
> >process some years back why not now complete the proceedure? Regarding
> >the DeCarle Spectrum analyzer board, the difficult chip has been the
> >RS232 driver. Bill in Canada has a supply of them and I see where most
> >mail order houses now carry the MAX equivelent chip which still fits the
> >boards you purchase four years ago!
> >
>
> Could you verify this URL please: I don't seem to be able to locate the host
> "w3ietc.ca"
>
> >Your next step is to check the home page of VE2IQ:
> >   http://w3ietc.ca/home/bill/spbill.htm
> >
>
> Thanks,
>
> --Johan http://w3ietc.ca/home/bill/spbill.htm does not have a vaild DNS entry
according to my server.

Tom

3. 56002 EVM

From: chbrain@dircon.co.uk (Charles Brain)
Date: Tue, 02 Apr 1996 19:26:01 +0000
To: hfsig@tapr.org
Message-ID: <199604021832.AA22613@felix.dircon.co.uk>
Rendered body type: text/plain

Well got my EVM56002 now, its the 66Mhz part !

Just need to find a box for it !

Regards Charles G4GUO

4. Re: [HFSIG:1008] 56002 EVM

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Tue, 2 Apr 1996 17:12:42 -0500
To: hfsig@tapr.org
Message-ID: <161A5E60.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

Way to go Charles!

     How much was it?

_____ Reply Separator _____
Subject: [HFSIG:1008] 56002 EVM
Author: hfsig@tapr.org at BALT.SMTP
Date: 4/2/96 12:37 PM

Well got my EVM56002 now, its the 66Mhz part !

Just need to find a box for it !

Regards Charles G4GUO

5. Re: [HFSIG:1009] Re: 56002 EVM

From: chbrain@dircon.co.uk (Charles Brain)
Date: Wed, 03 Apr 1996 06:53:39 +0000
To: hfsig@tapr.org
Message-ID: <199604030600.AA04943@felix.dircon.co.uk>
Rendered body type: text/plain

>   Way to go Charles!
>
>   How much was it?
>
>
>Well got my EVM56002 now, its the 66Mhz part !
>
>Just need to find a box for it !
>
>Regards Charles G4GUO
>
>
>
It cost me 99 U.K pounds + 8 pounds carriage and 17.5 % VAT. The U.S
price is slightly cheaper I believe.

CHB

6. Re: [HFSIG:1001] Re: Need simple PTT circuit

From: Mike Hooper <mhooper@netcom.com>
Date: Wed, 03 Apr 1996 16:00:43 -0800
To: hfsig@tapr.org
Message-ID: <1.5.4b12.32.19960404000043.00689b74@netcom.com>
Rendered body type: text/plain

At 10:49 AM 3/31/96 -0600, you wrote:
>Hello Mike,
>
>>From what I understood from Tom (and it was some long time ago), was that
>the time delay involved for the PTT command to work it's way from the DSP
>software, to the PC, to the PTT switch on the COM port, was unsatisfactory.
>That is because the sound card's DSP is doing HDLC and time slot persistance
>control and does not originate from the PC. Thus the delay involved and why
>he choose to add the PTT switch via the flag-out pin from where the DSP can
>directly control PTT. I'm not sure whether TOM's program stilll works like
>that.
>
>You may want to e-mail Tom and ask him for a solution. I agree that the COM
>port-derived PTT is the simplest way to go.
>
>Good luck, 73's

>
>--Johan

Thanks for your reply. Please send me Tom's email address if you have it. If I
develope a useful and reliable circuit or discover one I'll share it with the
group.

-73- Mike

Mike Hooper
mhooper@netcom.com
kf6pu@wb6ymh.#soca.ca.usa.noam

7. PC-DSP Software for Windows (Purchase Plan)

From: "James R. Kauten, M.D." <kauten@mindspring.com>
Date: Fri, 05 Apr 1996 00:00:37 -0500
To: dsp-93@tapr.org
Cc: hfsig@tapr.org
Message-ID: <199604050502.AAA16892@borg.mindspring.com>
Rendered body type: text/plain

For the past several weeks, the subject of digital signal processing has
been discussed on the TAPR DSP list serv. Software designed to facilitate
the further learning and modeling of DSP entities has also been discussed
(PC-DSP). Mention was made of a DSP course utilizing a text and a program
called PC-DSP. Thru discussion with PC Solutions, the developer of the
software, it was discovered that a Windows version is available. There
appears to be interest amongst members of the sig to purchase this software.

The web site for DSP Solutions is: http://www.dspsolutions.com. Samples of
PC-DSP for Windows and PC-SIM for Windows can be downloaded to sample.
Greg Jones, et.al. have made the suggestion that we inquire as to the
possibility of a bulk purchase discount.

PC Solutions is willing to give a volume discount, if there is enough
interest. The exact discount amount is unknown until the magnitude of the
order isdetermined.   It is possible that a group purchase could be made
thru TAPR, provided the discount is enough to defray the cost of processing
(thru TAPR) and yet provide a reasonable price to the purchasers.

Please send me your name if you are interested in purchasing either or both
of these programs. It should also be known that I have no monetary interest
in PC Solutions, my only interest is in obtaining the software at a
reasonable price for both myself and other hams who are interested.

If enough interest is generated, I will send out a summary as to the
discount and ordering details. If little interest is generated, I will let
those who responded know so they can place an order on their own.

73, Jim, KO4RQ
==========
James R. Kauten, M.D., KO4RQ
==========
Phone: (404) 355-9537 (Fax)   (404) 355-9515 (Daytime)
Address: James R. Kauten, M.D., 95 Collier Road, Suite 2055
                                          Atlanta, GA, 30305-1955
==========
Atlanta, Home of the 1996 Olympics!!!
==========

8. HF chan. simulator for 320C50

From: Peter Schulze TY1PS <ty1ps@bow.intnet.bj>
Date: Fri, 5 Apr 1996 09:40:23 +-200
To: "'HFSIG'" <hfsig@tapr.org>
Message-ID: <01BB22D3.F183A140@euraf>
Rendered body type: text/plain

Reading back the messages i see mention that the code for the HF channel
simulator is running on the TI 320C20. I have a TI module with the 320C50
now. I assume it works on that as well ? Is the code available and if so
where can i find it ?

Thanks
Peter TY1PS

9. TAPR DSP Web Pages

From: "Steven R. Bible" <srbible@mindport.net>
Date: Fri, 05 Apr 1996 12:14:07 -0500
To: dsp-93@tapr.org, hfsig@tapr.org
Message-ID: <1.5.4b12.32.19960405171407.00694788@mindport.net>
Rendered body type: text/plain

I have created what is a beginning of Digital Signal Processing web pages at the TAPR web server. Please point your browsers to:

    http://www.tapr.org/dsp/

THESE ARE YOUR PAGES and they are by no means finished. Presently there is information on the Alef Null DSP CARD 4 and Johan's EVM56K collection of programs. I have a text book bibliography and I've started on an articles bibliography. There are also pointers to other internet sources.

I need your help in making these pages what you want in DSP experimentation web pages. My goal is trying to organize everything there is concerning amateur radio DSP in one location. When I got started in DSP, I was overwelmed at the amount of information there was and how unorganized it is. That is what these pages are for. Remember when you got started in DSP? Remember how frustrating it was? Now is your chance to cahnge all this and organize all the material we can find on amateur DSP experimenting.

So here's what I would like to do.

- If you know of a DSP project, past, present or future, let's create an individual web page for each. Some projects I am aware of:

    - DSP-93
    - EVM56K (started)
    - TI DSK C5x
    - Alef Null DSP CARD (in progress)
    - any past and future projects

- Software. Organize software by project and function. Thus if you own a particular platform you can go to that platforms web page and see what's available. On the flip side, software organized by function so that if there is a partcular filter, modem, whatever you are interested in, you can review the source code for that function in attempts to learn or port over to another platform.

- Educational. We all want to learn more about DSP. Perhaps lessons can be posted here.

- any other ideas?

Please make suggestions and comments on the content, format, etc. that would make the pages work for you and others interested in DSP. If you have contributions to add, please send them to me, I'll add them. If you know of information that should be added, let me know, I'll add it. If you feel the pages would read better one way than another, let me know also. THESE ARE YOUR PAGES.

Onward!

- Steve, N7HPR

n7hpr@amsat.org
n7hpr@tapr.org

10. Re: [HFSIG:1013] HF chan. simulator for 320C50

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Fri, 05 Apr 1996 09:45:13 -0800
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19960405174513.3fef0b6e@ucs.orst.edu>
Rendered body type: text/plain

Hello Peter,

The simulator runs on several TI-based platforms such as the DG1SCR board of Juergen Hasch, either with a 320C25 or 320C26 chip installed. I have made a port to the DSP-93, which also is a TI320C25 (it took me many hours of fixes to get it to work). I am interested do some further work to run it on the TI 320C26 DSK just because then I can put it in a box to be used solely for that purpose.

I'm not aware of anyone working on a 320C50 version, but you may want to ask around - someone may well have done it already.

Alexander Kurpiers, DL8AAU, is the author of the software - it would be a
good idea to drop him a line. He told me that he was going to do some more
work to the simulator before releasing the source code

Are you interested in using something like the simulator? If there is
sufficient interest, I may be persueded to bring out a low-cost commercial
version - of course I'll write my own code for it - the principles are the same.

73's

--Johan

At 02:49 AM 4/5/96 -0600, you wrote:
>Reading back the messages i see mention that the code for the HF channel
>simulator is running on the TI 320C20. I have a TI module with the 320C50
>now. I assume it works on that as well ? Is the code available and if so
>where can i find it ?
>
>Thanks
>Peter TY1PS
>
>
>
>

11. Re: [HFSIG:1014] TAPR DSP Web Pages

From: Jim Sanford <JSANFORD@INFI.NET>
Date: Fri, 05 Apr 1996 19:33:38 -0500
To: hfsig@tapr.org
Message-ID: <199604060032.TAA26370@mh004.infi.net>
Rendered body type: text/plain

Steve
Great idea! Additions I'd like:
        DSP-12 pages. They're out of production, but good units, with
        many out there.
`   Algorithm pages. Let's share algorithms that can be ported anywhere
        and then share the code. This is a way to REALLY make some progress.

Incidentally, we have a mutual friend: KM4EM. He just finished the "Charm"
course and is enroute NR-1. Will hopefully be in touch by portable packet and
HF
73, Jim wb4gcs@amsat.org

At 11:31 AM 4/5/96 -0600, you wrote:
>I have created what is a beginning of Digital Signal Processing web pages at
>the TAPR web server. Please point your browsers to:
>
>   http://www.tapr.org/dsp/
>
>THESE ARE YOUR PAGES and they are by no means finished. Presently there is
>information on the Alef Null DSP CARD 4 and Johan's EVM56K collection of
>programs. I have a text book bibliography and I've started on an articles
>bibliography. There are also pointers to other internet sources.
>
>I need your help in making these pages what you want in DSP experimentation
>web pages. My goal is trying to organize everything there is concerning
>amateur radio DSP in one location. When I got started in DSP, I was
>overwelmed at the amount of information there was and how unorganized it is.
>That is what these pages are for. Remember when you got started in DSP?
>Remember how frustrating it was? Now is your chance to cahnge all this and
>organize all the material we can find on amateur DSP experimenting.
>
>So here's what I would like to do.
>
> - If you know of a DSP project, past, present or future, let's create an
>individual web page for each. Some projects I am aware of:
>
>   - DSP-93
>   - EVM56K (started)
>   - TI DSK C5x
>   - Alef Null DSP CARD (in progress)
>   - any past and future projects
>
> - Software. Organize software by project and function. Thus if you own a
>particular platform you can go to that platforms web page and see what's
>available. On the flip side, software organized by function so that if

>there is a partcular filter, modem, whatever you are interested in, you can
>review the source code for that function in attempts to learn or port over
>to another platform.
>
> - Educational. We all want to learn more about DSP. Perhaps lessons can
>be posted here.
>
> - any other ideas?
>
>Please make suggestions and comments on the content, format, etc. that would
>make the pages work for you and others interested in DSP. If you have
>contributions to add, please send them to me, I'll add them. If you know of
>information that should be added, let me know, I'll add it. If you feel the
>pages would read better one way than another, let me know also. THESE ARE
>YOUR PAGES.
>
>Onward!
>
>
> - Steve, N7HPR
>
>n7hpr@amsat.org
>n7hpr@tapr.org
>
>
>

12. Re: [HFSIG:1012] PC-DSP Software for Windows (Purchase Plan)

From: Jim Sanford <JSANFORD@INFI.NET>
Date: Fri, 05 Apr 1996 19:36:14 -0500
To: hfsig@tapr.org
Message-ID: <199604060035.TAA25912@mh004.infi.net>
Rendered body type: text/plain

What's the approximate price range????
73, Jim wb4gcs@amsat.org
At 11:14 PM 4/4/96 -0600, you wrote:
>For the past several weeks, the subject of digital signal processing has
>been discussed on the TAPR DSP list serv. Software designed to facilitate
>the further learning and modeling of DSP entities has also been discussed
>(PC-DSP). Mention was made of a DSP course utilizing a text and a program
>called PC-DSP. Thru discussion with PC Solutions, the developer of the
>software, it was discovered that a Windows version is available. There
>appears to be interest amongst members of the sig to purchase this software.
>
>The web site for DSP Solutions is: http://www.dspsolutions.com. Samples of
>PC-DSP for Windows and PC-SIM for Windows can be downloaded to sample.
>Greg Jones, et.al. have made the suggestion that we inquire as to the
>possibility of a bulk purchase discount.
>
>PC Solutions is willing to give a volume discount, if there is enough
>interest. The exact discount amount is unknown until the magnitude of the
>order isdetermined.   It is possible that a group purchase could be made
>thru TAPR, provided the discount is enough to defray the cost of processing
>(thru TAPR) and yet provide a reasonable price to the purchasers.
>
>Please send me your name if you are interested in purchasing either or both
>of these programs. It should also be known that I have no monetary interest
>in PC Solutions, my only interest is in obtaining the software at a
>reasonable price for both myself and other hams who are interested.
>
>If enough interest is generated, I will send out a summary as to the
>discount and ordering details. If little interest is generated, I will let
>those who responded know so they can place an order on their own.
>
>73, Jim, KO4RQ
>==
>James R. Kauten, M.D., KO4RQ
>==
>Phone: (404) 355-9537 (Fax)   (404) 355-9515 (Daytime)
>Address: James R. Kauten, M.D., 95 Collier Road, Suite 2055
>   Atlanta, GA, 30305-1955
>==
>Atlanta, Home of the 1996 Olympics!!!
>==
>
>
>

13. Re: [HFSIG:1017] Re: PC-DSP Software for Windows (Purchase Plan)

From: "James R. Kauten, M.D." <kauten@mindspring.com>
Date: Fri, 05 Apr 1996 20:40:17 -0500
To: hfsig@tapr.org
Cc: dsp-93@tapr.org
Message-ID: <199604060142.UAA11193@borg.mindspring.com>
Rendered body type: text/plain

At 06:50 PM 4/5/96 -0600, you wrote:
>What's the approximate price range????
>73, Jim wb4gcs@amsat.org
>At 11:14 PM 4/4/96 -0600, you wrote:
>>For the past several weeks, the subject of digital signal processing has
>>been discussed on the TAPR DSP list serv....
>>The web site for DSP Solutions is: http://www.dspsolutions.com...
>>PC Solutions is willing to give a volume discount,...

List (non discount) prices per PC Solutions are as follows:

PC-DSP for Windows $139.00
PC-SIM for Windows $159.00

Both $249.00
==========
James R. Kauten, M.D., KO4RQ
==========
Phone: (404) 355-9537 (Fax)   (404) 355-9515 (Daytime)
Address: James R. Kauten, M.D., 95 Collier Road, Suite 2055
                                          Atlanta, GA, 30305-1955
==========
Atlanta, Home of the 1996 Olympics!!!
==========

14. Re: [HFSIG:1016] Re: TAPR DSP Web Pages

From: "Steven R. Bible" <srbible@mindport.net>
Date: Sat, 06 Apr 1996 05:00:51 -0500
To: hfsig@tapr.org
Message-ID: <1.5.4b12.32.19960406100051.006a29bc@mindport.net>
Rendered body type: text/plain

At 06:48 PM 4/5/96 -0600, you wrote:
>Steve
>Great idea! Additions I'd like:
>   DSP-12 pages. They're out of production, but good units, with
>   many out there.
>`   Algorithm pages. Let's share algorithms that can be ported anywhere
>   and then share the code. This is a way to REALLY make some progress.
>
>Incidentally, we have a mutual friend: KM4EM. He just finished the "Charm"
>course and is enroute NR-1. Will hopefully be in touch by portable packet
and
>HF

Hello Jim, Chas has told me all about you (well, maybe not all:-). Perhaps
he's told you about me.

I just finished SOAC and am heading to the Pennsylvania BLUE as Strat Weps.
I head down to Kings Bay in one week. Chas and I missed each other by a few
weeks. Perhaps we'll run into each other on the East coast some time.

73,

 - Steve, N7HPR

n7hpr@amsat.org
n7hpr@tapr.org

15. [DSP-93:2127] PC-DSP/PC-SIM for Windows Order

From: "James R. Kauten, M.D." <kauten@mindspring.com>
Date: Sun, 07 Apr 1996 11:57:37 -0400
To: hfsig@tapr.org
Message-ID: <199604071557.LAA28926@borg.mindspring.com>
Rendered body type: text/plain

>Return-Path: dsp-93@tapr.org
>Date: Sat, 6 Apr 1996 21:58:24 -0600 (CST)
>Reply-To: dsp-93@tapr.org
>Originator: dsp-93@tapr.org

>Sender: dsp-93@tapr.org
>From: "James R. Kauten, M.D." <kauten@mindspring.com>
>To: dsp-93@tapr.org
>Subject: [DSP-93:2127] PC-DSP/PC-SIM for Windows Order
>X-Listprocessor-Version: 6.0 -- ListProcessor by Anastasios Kotsikonas
>X-Comment: Tucson Amateur Packet Radio DSP-93 Kit SIG
>
>For those interested in PC-DSP/PC-SIM for Windows:
>
>The process of obtaining quotes and a listing of those interested in the
>PC-DSP and PC-SIM software packages is progressing. Hopefully a notice will
>be published in the TAPR psr prior to Dayton. The article has been written.
>
> It is my hope that all can be finalized after Dayton. There are some
>TAPR/Digital forums at Dayton that may spark further interest, and an
>increase in the order may produce a better discount. Please be patient as
>this will take a little time to coordinate.
>
>73, Jim
>===
>James R. Kauten, M.D., KO4RQ (kauten@mindspring.com)
>===
>Phone: (404) 355-9537 (Fax)   (404) 355-9515 (Daytime)
>Address: James R. Kauten, M.D., 95 Collier Road, Suite 2055
>   Atlanta, GA, 30305-1955
>===
>Atlanta, Home of the 1996 Olympics!!!
>===
>
>
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James R. Kauten, M.D., KO4RQ (kauten@mindspring.com)
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Phone: (404) 355-9537 (Fax)   (404) 355-9515 (Daytime)
Address: James R. Kauten, M.D., 95 Collier Road, Suite 2055
                                              Atlanta, GA, 30305-1955
===
Atlanta, Home of the 1996 Olympics!!!
===

16. PACTOR discussed on radio show (fwd)

From: Greg Jones <wd5ivd@tapr.org>
Date: Tue, 9 Apr 1996 13:32:57 -0500 (CDT)
To: hfsig@tapr.org (HF SIG mailing)
Message-ID: <199604091832.NAA28653@tapr.org>
Rendered body type: text/plain

This from Len about his show this weekend.

We will probably have the show eventually on the TAPR pages in the coming
months if you don't have the chance to check the show live.

Cheers - Greg

Forwarded message:
Date: Tue, 09 Apr 1996 08:27:02 -0700
From: Len Winkler <lenwink@indirect.com>
Subject: PACTOR discussed on radio show

Paul Sussman, KB8LUJ, will be the special guest this sunday on the Ham
Radio & More radio show. Be sure to listen to find out more about
PACTOR. Show times and info appear below.
--
Len Winkler, KB7LPW   lenwink@indirect.com
P.O. Box 9219   kb7lpw@kc7y.az.usa.na
Phoenix, Az. 85068-9219
Ham Radio & More Show info at:
http://www.barc.org/barc/ham-more.html
RealAudio site:http://www.tapr.org/hrm/hrm.html

The show airs LIVE each SUNDAY at 6:00pm ET (2200utc) on many stations
throughout the country.
ALSO: LIVE everywhere on WWCR shortwave, 100,000 watts, on 7.435mhz and
12.160mhz, 2200utc.

The show also airs on WWCR shortwave, tape delayed at 0900utc on 3.315,
on Mondays, and Saturdays at 1600utc on 12.160.

Support "WOG". Written only General!!!

17. Progress PC-DSP/PC-SIM for Windows

From: "James R. Kauten, M.D." <kauten@mindspring.com>
Date: Tue, 09 Apr 1996 15:37:02 -0400
To: dsp-93@tapr.org
Cc: mullaneb@mccc.edu, xbrucex@popmail.mcs.net, 101365.1113@compuserve.com,
wd5ivd@tapr.org, hfsig@tapr.org, k4gvo@america.net, jsanford@infi.net,
kgoodwin@draper.com, johnson@mmsi.com, rbiasiot@nji.com,
lanier.r.a@smtpgty.bwi.wec.com, twatson@magic1.org, wpreez@csir.co.za,
ylshih@alumni.caltech.edu
Message-ID: <199604091936.PAA04059@borg.mindspring.com>
Rendered body type: text/plain

Have just received a quote from DSP Solutions as listed below. Presently we have approximately 12-15 people interested and hopefully this will increase once the notice is published in the TAPR PSR and after Dayton. I will keep you informed as to the progress. Greg Jones could perhaps look at these prices and determine whether it is feasible for these to be purchased thru TAPR. I would be willing to write them a check for my copy of the programs if it would keep processing cost down:

>Dear Dr. Kauten:
>
>Thank you again for your interest in DSP Solutions products. As I have
>indicated in my previous message, the single-copy price of the Windows
>bundle is $249. We can offer the following discounts for bulk purchases:
>
>Quantity Discount Price per copy
>
> 6-10 10% $ 224.10
>11-20 15% $ 211.65
>21-40 20% $ 199.20
>41-60 25% $ 186.75
>61 or more 30% $ 174.30
>
>We have not yet determined the pricing for developers' editions. PC-SIM
>developers' edition will be released shortly, and the PC-DSP developers'
>edition will be released about six weeks after that. The pricing will be
>determined at the time the software is released. We will offer special
>upgrade pricing for owners of standard editions. We can also offer
>discounts for bulk purchases similar to the above discount structure.
>
>Please let me know if I may provide more information.
>
>Sincerely,
>
>Oktay Alkin, Ph.D.
>DSP Solutions
>oalkin@dspsolutions.com

73, Tnx fer ur patience.

Jim, KO4RQ
==========
James R. Kauten, M.D., KO4RQ (kauten@mindspring.com)
==========
Phone: (404) 355-9537 (Fax) (404) 355-9515 (Daytime)
Address: James R. Kauten, M.D., 95 Collier Road, Suite 2055
Atlanta, GA, 30305-1955
==========
Atlanta, Home of the 1996 Olympics!!!
==========

18. TAPR Server Notice

From: Greg Jones <wd5ivd@tapr.org>
Date: Tue, 9 Apr 1996 18:05:07 -0500 (CDT)
To: tapr-bb@tapr.org (TAPR-BB mailing), hfsig@tapr.org (HF SIG mailing),
aprssig@tapr.org (BBS SIG mailing), netsig@tapr.org (NETSIG mailing),
bbssig@tapr.org (BBS SIG mailing), tapr-tnc@tapr.org (tapr-tnc)
Message-ID: <199604092305.SAA05153@tapr.org>
Rendered body type: text/plain

The TAPR server will be physically relocated during the weekend of 12 - 15 April. There may be times between now and Monday night when the system will be unavailable.

Tucson Amateur Packet Radio
8987-309 E Tanque Verde Rd #337 * Tucson, Az * 85749-9399 * 817-383-0000
---
e-mail: TAPR@TAPR.ORG ftp: ftp.tapr.org web: http://www.tapr.org/tapr
---

19. Page 4 of DSp Card 4 User Manual

From: chbrain@dircon.co.uk (Charles Brain)
Date: Mon, 15 Apr 1996 22:31:38 +0100
To: hfsig@tapr.org
Message-ID: <199604152140.AA29238@felix.dircon.co.uk>
Rendered body type: text/plain

Help,

I am trying to calculate the coefficients of of two FIR filter whose
outputs are 90 out of phase. I am trying to produce the series from

hi(n)=2h(n)cos(wc*n)
hq(n)=2h(n)sin(wc*n)

I am using the formula

cos(2*pi*frequency/sample_frequency). To calculate the series cos(wc*n),
i.e the second term.

When I multiply this with the BPF I have generated I don't get anything like
Fig 23, in fact I get a band reject filter.

I have obviously done something silly but what ? I can't any reference to this
technique in the DSP books I have ?

73 Charles

20. TAPR.ORG Status

From: Greg Jones <wd5ivd@tapr.org>
Date: Tue, 16 Apr 1996 19:38:19 -0500 (CDT)
To: tapr-bb@tapr.org (TAPR-BB mailing), aprssig@tapr.org (BBS SIG mailing),
        netsig@tapr.org (NETSIG mailing), bbssig@tapr.org (BBS SIG mailing),
        hfsig@tapr.org (HF SIG mailing), dsp-93@tapr.org (DSP-93 Build),
        das@tapr.org (das), tapr-tnc@tapr.org (tapr-tnc)
Message-ID: <199604170038.TAA13797@tapr.org>
Rendered body type: text/plain

As your e-mail boxes show, TAPR.ORG is again connected to the net.
DataRace, our hosts, relocated their offices this weekend which caused the
system outage. While TAPR.ORG was only down for a short time, without the
connection to the service provider it was hard to service connections.

If you sent mail during the weekend, there is a good chance that your mail
provider held your message and sent it when our connection was
reestablished. If you don't see your e-mail then just send it again.

There will be some additional small outages during the week as the MIS folks
at DataRace clean up the connection and move wires around.

Thanks for everyone patients during this down time.

Cheers - Greg, WD5IVD

21. RE: [HFSIG:1024] Page 4 of DSp Card 4 User Manual

From: Jon Bloom <jbloom@nai.net>
Date: Tue, 16 Apr 1996 22:56:07 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB2BE7.E79A27E0@naihp52.nai.net>
Rendered body type: text/plain

Charles Brain[SMTP:chbrain@dircon.co.uk] wrote:
>I am trying to calculate the coefficients of of two FIR filter whose
>outputs are 90 out of phase. I am trying to produce the series from
>
>hi(n)=2h(n)cos(wc*n)
>hq(n)=2h(n)sin(wc*n)
>
>I am using the formula
>
>cos(2*pi*frequency/sample_frequency). To calculate the series cos(wc*n),
>i.e the second term.
>

>When I multiply this with the BPF I have generated I don't get anything like
>Fig 23, in fact I get a band reject filter.

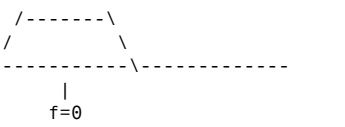
Yes, I bet you do!

>I have obviously done something silly but what ?

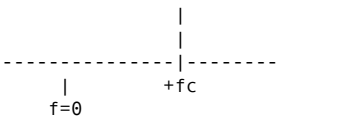
Well, you started with a band-pass filter, for one thing. You need to start
with a *low-pass* filter. Let me see if I can explain why.

What this technique does is take a signal--which is actually the filter's
impulse response--and mix it with a sinusoid. This is mixing, just like in
a receiver. It produces sum and difference frequencies. A better way to
look at it though, I think, is to look at the process as a frequency shift.
Multiplying a signal by a sinusoid causes the signal to shift in frequency.
Here's the frequency-domain situation if the h(n) is the impulse response of
a low-pass filter, rendered in crude ASCII:

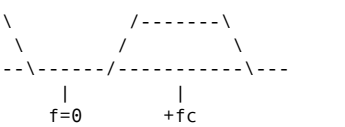
LPF frequency response:



Note that I am showing both the positive and negative frequencies. Now we
multiply this signal by the sinusoid, which has a spectrum like so:



It, too, has positive and negative frequency components. The positive
frequency component of the sinusoid shifts the filter response up in
frequency by fc, while the negative frequency component shifts the filter
response down by fc, resulting in the following filter-response spectrum:



Note that this is just the response of a bandpass filter centered
at fc.

Now if you do this twice, once using a sinusoid of, say, 0 degrees
phase as the fc signal, and a second time using a sinusoid of 90
degrees phase for fc, you end up with two bandpass filters that have
magnitude responses that are identical but phase responses that
differ by 90 degrees. Which is what you were looking for in the
first place!

A couple of things should be clear from these pictures (if anything
is clear at all at this point!) First, the bandwidth of the BPFs
that result from this operation is twice the bandwidth of the
original LPF. Second, you need to watch the high end of the BPF,
to ensure that it doesn't get shifted up above fs/2 (fs is the
sampling frequency), to avoid aliasing of the filter response.
Likewise, you need to use a high enough fc to ensure that the left
(negative) side of the LPF response moves fully into the positive
side of the frequency spectrum.

Often, you might use fs/4 as the frequency for fc. In this case,
the original LPF must attenuate all frequencies above fs/4, because
when you shift the LPF up by fs/4, the original fs/4 response
becomes the fs/2 response.

By the way, fs/4 is a handy fc to use because all of the
coefficients of that signal (for 0 and 90 phases) are 0, 1 or -1.

With all that in mind, see if the result of what you did by shifting
a BPF now makes sense, bearing in mind that both the positive-frequency
response *and* the negative-frequency response of the filter get
shifted, and they get shifted both up and down.

So in brief, here's the correct procedure:

1) Determine the desired bandwidth and center frequency of the BPF.

2) Design a LPF that has a bandwidth 1/2 the desired BPF bandwidth.

3) Generate quadrature (sin and cos) signals at a frequency (fc) equal
to the center frequency of the desired BPF. i.e:

```
       i(n) = sin(2*pi*fc/fs)
     q(n) = cos(2*pi*fc/fs)
```

4) Generate the BPF impulse responses by:

```
     hi(n) = 2h(n)i(n)
     hq(n) = 2h(n)q(n)
```

where h(n) is the impulse response of the LPF. (The factor of 2 occurs
because the original filter bandwidth is getting doubled in the process,
so the passband amplitude response of the filter is halved.)

>I can't any reference to this
>technique in the DSP books I have ?

Well, you need better books. One I know of that explains this
process in detail is "Digital Signal Processing in Communication
Systems," by Marvin E. Frerking, Van Nostrand Reinhold, New York,
1994. (ISBN 0-442-01616-6)

-- Jon, KE3Z

22. Re: [HFSIG:1024] Page 4 of DSp Card 4 User Manual

From: Jon Bloom <jbloom@nai.net>
Date: Wed, 17 Apr 1996 07:14:38 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB2C2D.8DF1B8C0@naihp22.nai.net>
Rendered body type: text/plain

I think I sent this with Exchange encapsulation turned on (again), so
I'm resending it.

Charles Brain[SMTP:chbrain@dircon.co.uk] wrote:
>I am trying to calculate the coefficients of of two FIR filter whose
>outputs are 90 out of phase. I am trying to produce the series from
>
>hi(n)=2h(n)cos(wc*n)
>hq(n)=2h(n)sin(wc*n)
>
>I am using the formula
>
>cos(2*pi*frequency/sample_frequency). To calculate the series cos(wc*n),
>i.e the second term.
>
>When I multiply this with the BPF I have generated I don't get anything like
>Fig 23, in fact I get a band reject filter.

Yes, I bet you do!

>I have obviously done something silly but what ?

Well, you started with a band-pass filter, for one thing. You need to start
with a *low-pass* filter. Let me see if I can explain why.

What this technique does is take a signal--which is actually the filter's
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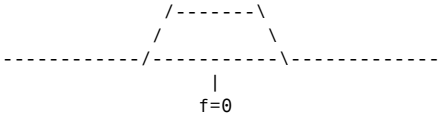
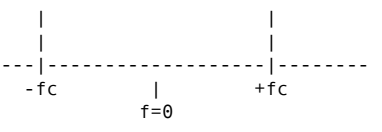


Note that I am showing both the positive and negative frequencies. Now we
multiply this signal by the sinusoid, which has a spectrum like so:



It, too, has positive and negative frequency components. The positive
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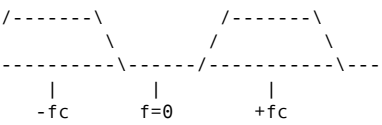
frequency by fc, while the negative frequency component shifts the filter
response down by fc, resulting in the following filter-response spectrum:



Note that this is just the response of a bandpass filter centered
at fc.

Now if you do this twice, once using a sinusoid of, say, 0 degrees
phase as the fc signal, and a second time using a sinusoid of 90
degrees phase for fc, you end up with two bandpass filters that have
magnitude responses that are identical but phase responses that
differ by 90 degrees. Which is what you were looking for in the
first place!

A couple of things should be clear from these pictures (if anything
is clear at all at this point!) First, the bandwidth of the BPFs
that result from this operation is twice the bandwidth of the
original LPF. Second, you need to watch the high end of the BPF,
to ensure that it doesn't get shifted up above fs/2 (fs is the
sampling frequency), to avoid aliasing of the filter response.
Likewise, you need to use a high enough fc to ensure that the left
(negative) side of the LPF response moves fully into the positive
side of the frequency spectrum.

Often, you might use fs/4 as the frequency for fc. In this case,
the original LPF must attenuate all frequencies above fs/4, because
when you shift the LPF up by fs/4, the original fs/4 response
becomes the fs/2 response.

By the way, fs/4 is a handy fc to use because all of the
coefficients of that signal (for 0 and 90 phases) are 0, 1 or -1.

With all that in mind, see if the result of what you did by shifting
a BPF now makes sense, bearing in mind that both the positive-frequency
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to the center frequency of the desired BPF. i.e:

```
        i(n) = sin(2*pi*fc/fs)
      q(n) = cos(2*pi*fc/fs)
```

4) Generate the BPF impulse responses by:

```
      hi(n) = 2h(n)i(n)
      hq(n) = 2h(n)q(n)
```

where h(n) is the impulse response of the LPF. (The factor of 2 occurs
because the original filter bandwidth is getting doubled in the process,
so the passband amplitude response of the filter is halved.)

>I can't any reference to this
>technique in the DSP books I have ?

Well, you need better books. One I know of that explains this
process in detail is "Digital Signal Processing in Communication

Systems," by Marvin E. Frerking, Van Nostrand Reinhold, New York,
1994. (ISBN 0-442-01616-6)

-- Jon, KE3Z

23. Re: [HFSIG:1024] Page 4 of DSp Card 4 User Manual

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Wed, 17 Apr 1996 09:49:57 -0800
To: hfsig@tapr.org
Message-ID: <11F75CC608E@frl.orst.edu>
Rendered body type: text/plain

Hi Charles,

I think Jon gave a pretty neat explanation of how to develop the 90 degree
filters.

Here is how Pawel, SP9VRC, does this I/Q splitting in his modems. First he
decided on the window shape, i.e., in his case he wanted to use
the three-term Blackman-Harris window for low sidelobe leakage. The rest is
all generated by the assembler for you. Things are very nicely
parameterized for sample rate, passband and filter order. If you follow
this bit of code, it becomes a template for many other applications. You
don't need to compute filter coefficients by hand every time.

This is one lesson that I learned the hard way - if you are serious about
doing DSP work, invest in a decent assembler that provides good macro
support. The assembler that comes with the DSP56KEVM not only provides
for macros, but also provides very useful directives as shown in the code
example.

My two-cents for what it's worth.

GL and 73

--Johan

```
;*** TAKEN FROM PAWEL JALOCHA'S (SP9VRC) CODE EXAMPLES *****************
;
;
;constant tables: FIR and window shapes

LowFreq equ AliasLowFreq/SampleFreq
UppFreq equ AliasUppFreq/SampleFreq

w0      equ     0.35875 ;coeff. for Blackman-Harris 4 term
w1      equ     0.48829 ;minimum sidelobe window (copied from Motorola BBS)
w2      equ     0.14128
w3      equ     0.01168

        org Y:AliasFilterI

time = -@cvf(AliasFilterLen/2)+0.5
count   set 0
        dup AliasFilterLen
angle = PI*time/@cvf(AliasFilterLen/2)
Window = w0+w1*@cos(angle)+w2*@cos(2.0*angle)+w3*@cos(3.0*angle)
Filter = (@sin(2.0*PI*time*UppFreq)-@sin(2.0*PI*time*LowFreq))/(PI*time)
time = time+1.0
        dc  Window*Filter
count   set count+1
        endm

        org Y:AliasFilterQ

time = -@cvf(AliasFilterLen/2)+0.5
count   set 0
        dup AliasFilterLen
angle = PI*time/@cvf(AliasFilterLen/2)
Window = w0+w1*@cos(angle)+w2*@cos(2.0*angle)+w3*@cos(3.0*angle)
Filter = (-@cos(2.0*PI*time*UppFreq)+@cos(2.0*PI*time*LowFreq))/(PI*time)
time = time+1.0
        dc  Window*Filter
count   set count+1
        endm
*************************************************************************
```

24. Phase shift (Re: Page 4 of DSp Card 4 User Manual)

From: "Eric E. Silbaugh" <silbaugh@apci.net>
Date: Wed, 17 Apr 1996 22:05:16 -0500
To: chbrain@dircon.co.uk (Charles Brain)
Cc: hfsig@tapr.org
Message-ID: <199604180305.WAA32362@hilly.apci.net>
Rendered body type: text/plain

Charles,

I don't have the manual you mention, so I don't know what Fig. 23 shows. It seems to me that a band-reject filter is exactly what you should get. Obviously, that's not what you expected! Here's what seems to have happened.

Since you are trying to produce signals with a 90 degree phase shift (and probably no other change) I presume your band-pass filter, h(n), has a fairly wide response. By multiplying the filter coeficients with a sinusoid, cos(wc*n) or sin(wc*n), you modulated the filter frequency response up and down in frequency. This is exactly what happens in an analog system (sum and difference frequencies).

I presume the frequency of your modulating sinusoids was less than Fs/2. If this is the case, then the filter frequency response which was modulated down in frequency appears around DC (zero Hz). The notch appears because your original band-pass filter had nulls near DC and Fs/2. I'll predict that the notch appears near the frequency of the modulating sinusoid.

If the sinusoid frequency was above Fs/2 you aliased part of the filter frequency response which was near Fs down to DC. Remember, in sampled (digital) systems the spectrum repeats at intervals of the sampling frequency, Fs. In this case, I won't try to guess where the null appeared. The why of the null has the same explanation as before.

Now for the good news. You have almost reinvented the Weaver method for generating SSB! You can generate your in-phase (I) and quadrature (Q) signals by eliminating the final two RF mixers in the Weaver method. Here's what to do:

1. Make your filter, H(n), a low-pass filter instead of a band-pass.

2. Modulate your digital signal instead of the filter.

3. Send your modulated signal through the low-pass filters and the outputs are the desired I and Q signals.

In equations,

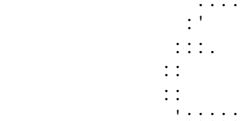
```
   I(n) = M(n)*cos(wc*n)**H(n)
   Q(n) = M(n)*sin(wc*n)**H(n)
```

where M(n) is the sampled message signal (voice, data), H(n) is a low-pass filter, cos(wc*n) and sin(wc*n) are the modulating sinusoids, and ** is the convolution operator (that's the mathematical operation you perform when you filter a signal).

You should be able to find an explanation of the Weaver method in any good communications textbook (it may even be in your manual). Brush up on your Fourier theory (or learn it for the first time) and really study the explanation of the Weaver method using analog signals. You can then translate this to your digital system. Understanding what happens to analog signals in the frequency domain will help you choose the modulating frequency of the sinusoids and the low-pass filter cutoff frequency in your digital system. Never forget, in sampled (digital) systems the spectrum repeats at intervals of the sampling frequency, Fs.

Hope this helps.

Eric, N2NNP



Eric E. Silbaugh
N2NNP
silbaugh@apci.net

25. Re: Phase shift

From: "Eric E. Silbaugh" <silbaugh@apci.net>
Date: Wed, 17 Apr 1996 23:45:37 -0500

To: chbrain@dircon.co.uk (Charles Brain)
Cc: Jon Bloom <jbloom@nai.net>, hfsig@tapr.org
Message-ID: <199604180445.XAA04432@hilly.apci.net>
Rendered body type: text/plain

Charles,

Just minutes after I sent my reply to you I received an HFSIG digest with Jon's, KE3Z, reply to your message. I realized I may have just completely confused you.

If you follow the procedure I described you may not get the signal you were trying to generate. You will get two signals whose positive and negative frequency parts will have the same phase relationship as those produced by the method Jon described (90 degree phase shift). However, the signals will not be the same as those produced by Jon's method.

I had to sit down and actually do the math to prove to myself that the phase shifts produced by these two methods are actually identical. It really works. The signals are different, the filters are different, but the phase shifts are identical.

What all this means is that, in theory, you can use either set of signals to do identical signal processing operations. For example, either set could be used in an I/Q modulator to produce SSB. Weird, using different signals to get the same result. Sometimes the universe really is stranger than we can imagine.

Please note the emphasis on the phrase 'in theory' in the above. I'm sure there are some 'gotchas' just lying in wait for the unwary experimenter. Now, what are the pros and cons to each method? Is one method more efficient to implement than the other? What about the flexibility in choosing modulation frequencies and filter bandwidths?

Jon's comments on using Fs/4 for the sinusoidal modulating frequency also apply to the method I described. If you do this you can reduce the number of computations needed for the filters by half.

I'm sure that's plenty for now.

Eric


```
    .:::.  .:::.  .:::.
   :'     :'     :'
  :::.   :::.    ':.           Eric E. Silbaugh
 ::     ::          ':.             N2NNP
 ::     ::           ::        silbaugh@apci.net
  ':::::::::::::::::'
```


26. $99 Floating Point DSK

From: bobh@inav.net
Date: Thu, 18 Apr 1996 16:35:18 -0500
To: hfsig@tapr.org
Message-ID: <3176B596.4137@inav.net>
Rendered body type: text/plain

Another DSP evaluation kit. This one is based on TI's floating point TMS320C31.

No mention of software (development or applications) that might be included.

Prices are so low. I soon expect to see mass mailings of DSK boards much like that with AOL diskettes.

wm6h

==========

The 'C3x DSK is a low-cost, easy-to-use, high performance, and expandable development platform that lets you experiment and develop real-time signal processing applications with the TMS320C3x DSP. The DSK has a TMS320C31 on board
to allow full-speed verification of the TMS320C3x code. The DSK also gives you
the freedom to develop your own software on a PC and run it on the DSK board. Moreover, the DSK allows you to expand your system by building new daughterboards. The DSK interfaces to a host PC through the parallel printer port.

More than 15 third-parties have expressed interest in developing modules for

the
'C3X DSK. The 'C3X DSK, will be available through authorized TI distributors
for
$99 on April 15, 1996.

The following sections describe the key features of the 'C3x DSK.

Key Features

TMS320C31 floating point DSP
40 ns instruction cycle (50MFLOPS, 25 MIPS)
Standard or Enhanced Parallel Printer Port Interface
Voice quality analog data acquisition via the TLC32040 analog interface
circuit:
14-bit dynamic range ADC and DAC
variable ADC and DAC sampling rate up to 20,000 samples per second output
reconstruction filter and bypassable switched-capacitor
antialiasing input filter
Standard RCA plug connectors for analog input and output that provide a
direct
connection to microphone and speaker
XDS 510 emulator connector
I/O expansion bus for daughterboards

==========

27. Re: [HFSIG:984] Updated HF Channel Simulator

From: chbrain@dircon.co.uk (Charles Brain)
Date: Mon, 22 Apr 1996 18:13:20 +0100
To: hfsig@tapr.org
Message-ID: <199604221723.AA01048@felix.dircon.co.uk>
Rendered body type: text/plain

>Hi,
>
>For those interested in the DSP-93 version of the HF channel simulator,
>please check out the latest upload of an experimental HF channel simulator
>test.
>
>With Alexander's (DL8AAU) help, I managed to fix a few errors, and also
>included some new test conditions. The examples now reflect the CCIR
>specifications more accurately and now include:
>
>Selective fading: Good, moderate, poor, flutter fading,
>two kinds of flat fading: Good and severe.
>
>The updated package is available for downloading from:
>ftp.tapr.org/tapr/SIG/HFSIG/upload/simtest1.zip
>
>Trust you find it useful.
>
>73's
>
>--Johan Forrer, KC7WW
>
>
Hello Johan,
 Just a public thankyou for your and Alexanders work. I have it up and
running and have piped my 8 ary modem though it.
Guess what with no windowing of the input to the FFT, the modem wont
even sync, add a 4ms guard and it works again !

I look forward to noise being added.

73 Charles G4GUO

28. Re: [HFSIG:1032] Re: Updated HF Channel Simulator

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Mon, 22 Apr 1996 14:42:12 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960422143213.14880A-100000@ucs.orst.edu>
Rendered body type: text/plain

On Mon, 22 Apr 1996, Charles Brain wrote:

>
> Hello Johan,
> Just a public thankyou for your and Alexanders work. I have it up and
> running and have piped my 8 ary modem though it.

> Guess what with no windowing of the input to the FFT, the modem wont
> even sync, add a 4ms guard and it works again !
>
> I look forward to noise being added.
>
> 73 Charles G4GUO
>
>

Hi Charles,

See you made good use of the simulator. Which test conditions are you
using? Also wonder whether the modem works under all the test conditions?

We should thank Alexander and Juergen for all their hard work.
As far as I know, the noise generator is already there, so is the input
power measurement algorithm - I'm sure it would not take a lot to get
that working and allow Eb/N0 comparisons to be made.

73's

--Johan, KC7WW

29. Golay (24,12) extended code

From: Richard Lay <lay@cod.nosc.mil>
Date: Mon, 22 Apr 96 15:21:24 PDT
To: hfsig@tapr.org
Message-ID: <9604222221.AA27032@marlin.nosc.mil>
Rendered body type: text/plain

I've read somewhere that the Golay (24,12) extended error correcting code
(ecc) can always correct up to 3 errors, sometimes correct 4 errors, and
never correct 5 errors. Does anyone know the probability that this ecc will
correct 4 errors? What does this probability depend on?

Thanks,

Rich

30. SSB Filters

From: Pete Hardie <hardie@duke.usask.ca>
Date: Mon, 22 Apr 1996 18:44:35 -0600 (CST)
To: "HFSIG List - TAPR.ORG" <hfsig@tapr.org>
Message-ID: <Pine.OSF.3.91.960422183859.28941D-100000@duke.usask.ca>
Rendered body type: text/plain

I've just received an ADSP EZ-KIT Lite and have diddled its FIRDEMO
program so that it samples at 5.5125kHz (instead of 8kHz) and
replaced three of its filters with 100, 250 and 500Hz CW filters.
I'd like to add a couple of SSB filters but I'm not sure what the
bandpass characteristics of an SSB filter should be. It's obvious with a
CW filter what the bandpass edge frequencies are, but what are the
bandpass edges for a typical SSB filter? When designing the narrow SSB
filters do you reduce the high end or the low end, or squeeze them together?

Thanks.

73 de Pete
ve5va.qrp@usask.ca

31. RE: [HFSIG:1035] SSB Filters

From: Stephan Sykes <ssykes@ns2.emirates.net.ae>
Date: Tue, 23 Apr 1996 06:33:11 +-400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB30DE.FC826EA0@csa062.emirates.net.ae>
Rendered body type: text/plain

The normal bandpass for SSB voice is 300 to 3000 hz. In the dsp filters of our tactical r/t we reduce
the high end.

steve sykes
kd2om/a61aa

From: Pete Hardie[SMTP:hardie@duke.usask.ca]
Sent: Tuesday, April 23, 1996 0:01 AM
To: hfsig@tapr.org
Subject: [HFSIG:1035] SSB Filters

I've just received an ADSP EZ-KIT Lite and have diddled its FIRDEMO
program so that it samples at 5.5125kHz (instead of 8kHz) and
replaced three of its filters with 100, 250 and 500Hz CW filters.
I'd like to add a couple of SSB filters but I'm not sure what the
bandpass characteristics of an SSB filter should be. It's obvious with a
CW filter what the bandpass edge frequencies are, but what are the
bandpass edges for a typical SSB filter? When designing the narrow SSB
filters do you reduce the high end or the low end, or squeeze them together?

Thanks.

73 de Pete
ve5va.qrp@usask.ca

begin 600 WINMAIL.DAT
M>)\^(C@"`0:0" `$`````````!``$``0>0!@`(````Y 0```````#H``$-@ 0`
M`@````(``@`!!) &```!```!````# ````,``# #````"P`/#@`````"`?\/
M`0```#L`````````@2L?I+ZC$!F=;@#=`0]4`@````!H9G-I9T!T87!R+F]R
M9P!33510`&AF<VEG0'1A<'(N;W)G```>``(P`0````4```!33510`````!X`
M`S !````#P```&AF<VEG0'1A<'(N;W)G```#`!4,`0````,`_@\&````'@`!
M, $````1````)VAF<VEG0'1A<'(N;W)G)P`````"``0LP`0```!0```!33510
M.DA&4TE'0%1!4%(N3U)'``,``#D`````"P! .@$````"`?8/`0````0`````
M```#P2D!"(``!@```!)4$TN36EC<F]S;V9T($UA:6PN3F]T90`Q" $$@ $`
M'0```%)%.B!;2$9324<Z,3 S-5T@4U-"($9I;'1E<G,`'@@!!8 #``X```#,
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M;S>],+L!'@!P``$````=````4D4Z(%M(1E-)1SHQ,#,U72!34T(@1FEL=&5R
M<P``````"`7$``0```!8````!NS"]-UY/M&J!G \1S["Y1$535 ``````>`!X,
M`0````4```!33510`````!X``PP!````%P```'-S>6ME<T!E;6ER871E<RYN
M970N864```,`!A# ?@%$`P``$,("```>``@0`0```&4```!42$5.3U)-04Q"
M04Y4%34T9/4E-30E9/24-%25,S,#!43S,P,#!(6DE.5$A%1%-01DE,5$52
M4T]&3U525$%#5$E#04Q2+U1715)%1%5#151(14A)1TA%3D135$5615-92T53
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M+B%&0"$0(Y N=7,;\&OZ+B! 72F/*IT&8 (P*\\%+-M4"E!S9&%Y+"<4L!-0
M`Q$R,S20,3E .38@,#HP&M!!9DTP#RJ=5&\R3RS;:%L#T"&P0 &0$U N!;!G
MXS8?,1YU8FHHH3@_+-N 6TA&4TE'.ADQ=#5='&-&'Q0D[R7S,TXV)V<:12AF
M22<C06HO+W %0!80'.!I(T!D($#D4%$4U @15J +4M)5"!,:2W!&QNQ'?!A
M(T$N(&1D; =$P47P!"!&25)$19Q-3R*6*&$)P&%M(P#['9 >@&$%0$7P(P!(
MP M00P>124$U+C4Q&$!KL$AZ("@+@!/!8430]1^!.$JQ*48C"H46$ M1_QS@
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M7R8@(Y =<DM01-$@!:!USTG1'W(<<A\&8G5#HDC0[QLP28$(<!L0=TDR'H(*
MA?\;IQ%R($&<1/ '- ?8P.1JU,((P!H"&!L1-!B+U")'D!T)Q]A8G9I"&!_
M3H520 J%4"=4R!NG"8!GDQL0`U!E<0I0;F,(D/]*`180-)!3PE337:%5'UQ6
M8QP44D!T>7 @,U,(/_P@5QL``Z -L#HQ`P`9$/T><VX*P -@!^ <<@J%'P;Z
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M%3$`<: ```,`$! ``````P`1$ ````! ``<PX&,9M[PPNP% ``@PX&,9M[PP
8NP$>`#T``0````4```!213H@`````%IP
`
end

June 1996

Original source file: hfsig.9606
Source: https://archive.org/details/tapr-1997
Recovered from: https://archive.org/download/TAPR-HF_1996-06/TAPR-HF_1996-06.txt

1. QPSK BANDWIDTH ?

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Sun, 02 Jun 1996 21:33:32 -0800
To: HFSIG@tapr.org
Message-ID: <1.5.4.16.19960603053332.29172514@ucs.orst.edu>
Rendered body type: text/plain

Hi all,

Could anyone please comment on the expected bandwidth for a DQPSK signal at
T bauds (or 2*T bits/sec) with raised cosine pulse shaping.

I have been monitoring my QPSK signal using an FFT analyser program. My
modulator uses a root raised cosine response with x/sinx compensation with
alpha rolloff factor 0.2. While the QPSK signal is cycling though all its
possible bit patterns, I observe five peaks in the emitted spectrum as follows:

carrier at   0 dB,
+/- half T at -6 dB,
+-   T at -14 dB.

Everything else is way down in the noise. From this observation, it appears
that the signal is contained into T Herz. Is this what is supposed to be
there? I am a little puzzled by the two components at +/- half T, also not
too clear whether I should or not use the x/sinx compensation in this case
and whether the rolloff factor is suitable. Theoretically, it looks OK on
simulations - The signal sounds nice and clean and melodic, but I may be
missing something.

Any comments and suggestions will be appreciated.

--Johan

2. [HFSIG:1165] QPSK BANDWIDTH ? -Reply

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Mon, 03 Jun 1996 20:16:15 +0200
To: hfsig@tapr.org
Message-ID: <s1b2a086.016@KIDD.CO.ZA>
Rendered body type: text/plain

Hi All.

I have made some pic's (jpg format) of my EVM interface. Interesting
persons can email me for copies otherwise I will ask Johan to upload it to
the web page for hf signal processing

73 danie
BRYD@KIDD.CO.ZA

3. Re: [HFSIG:1165] QPSK BANDWIDTH ?

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)
Date: Mon, 3 Jun 96 08:17:47 CDT
To: hfsig@tapr.org
Message-ID: <9606031317.AA02940@eagle.aud.alcatel.com>
Rendered body type: text/plain

> Hi all,
>
> Could anyone please comment on the expected bandwidth for a DQPSK signal at
> T bauds (or 2*T bits/sec) with raised cosine pulse shaping.
>
> I have been monitoring my QPSK signal using an FFT analyser program. My
> modulator uses a root raised cosine response with x/sinx compensation with
> alpha rolloff factor 0.2. While the QPSK signal is cycling though all its
> possible bit patterns, I observe five peaks in the emitted spectrum as follows:
>
> carrier at   0 dB,
> +/- half T at -6 dB,
> +-   T at -14 dB.
>
> Everything else is way down in the noise. From this observation, it appears

> that the signal is contained into T Herz. Is this what is supposed to be
> there? I am a little puzzled by the two components at +/- half T, also not
> too clear whether I should or not use the x/sinx compensation in this case
> and whether the rolloff factor is suitable. Theoretically, it looks OK on
> simulations - The signal sounds nice and clean and melodic, but I may be
> missing something.
>
> Any comments and suggestions will be appreciated.
>
> --Johan
>

Johan: it is important that all measurements be done on a linear channel -- ie: a class-C amplifier will significantly modify the spectrum of a DQPSK signal. A staggered-offset-QPSK (OQPSK, or SQPSK) signal will not be widened as badly as straight QPSK, but both will widen when limited or clipped.

Secondly, it is important that a PRBS (of sufficient length) be used to excite the filter, else you may get individual spectral lines showing through. If you have a repetitive pattern, the energy at the discrete spectral lines may confuse things. If your FFT is of short length, and thus it truncates the PRBS pattern, this may also lead to measurement error. Additionally, windowing of the FFT, and compensation of amplitude induced errors of the window may prove useful.

I beleive that x/sinx compensation is correct if you are filtering square-wavish pulses. If you are filtering impulses, then compensation is of course not necessary.

You should have -3 dB at the +/- half T points of the TX output spectrum regardless of the alpha factor, for a root-cosine transfer function that has been properly compensated. It then falls off faster past +/- 1/2T. At +/- T there should be no output (filtered or not). However, it is common to have spectral leakage at +/- 1/T due to feedthrough of the baud clock, and some effort may be needed to track this down. An alpha factor of 0.2 leads to some pretty wild ringing, and it's important to assure the dynamic range of everything is preserved. Also, a sufficient number of filter taps is needed to have good out-of-band rejection at an alpha factor of 0.2, I think about 33 taps were needed for my prototype in Excel to have adequate out of band rejection.

It sounds as though you are making excellent progress, and am glad to see you using root-cosine filters so that the emitted spectrum will be narrow. The root-cosine filter on receive will help assure good noise bandwidth of the receiver. Incidentally, the noise bandwidth of a root-cosine filter is independent of the alpha factor.

- Tom, N5EG

Tom McDermott | "All opinions expressed
Alcatel Network Systems, Inc. | are my own, and do not
mcdermot@aud.alcatel.com | represent those of Alcatel
[ICC'96 Technical Program Secretary] | Network Systems, Inc."
[June 23-27, 1996, Dallas, Tx.] |

4. Center freq for rtty ?

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Tue, 04 Jun 1996 03:27:36 +0200
To: hfsig@tapr.org
Message-ID: <s1b3057d.004@KIDD.CO.ZA>
Rendered body type: text/plain

By the way...

Why was such high tones selected for RTTY ? (2125Hz mark, 2295Hz space) My hf rig shows much better response at 800 to 1000Hz. Would it not be better to implement the channel filters for a RTTY DSP modem around say 800Hz ?

I am busy with a rtty dsp modem for the evm and is now wondering about some of my earlier choices :-) From a DSP point of view would it not be better to work around 800Hz ? I have used my PK232 with great success in the past using my 500Hz CW filter on RTTY.

danie zs6awk

5. Re: [HFSIG:1168] Center freq for rtty ?

From: Pete Hardie <hardie@duke.usask.ca>
Date: Mon, 3 Jun 1996 10:36:38 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.93.960603102921.3180A-100000@duke.usask.ca>
Rendered body type: text/plain

On Mon, 3 Jun 1996, Danie Brynard wrote:
> Why was such high tones selected for RTTY ? (2125Hz mark, 2295Hz
> space)

Back when I did a little RTTY with my KAM, I found that it could copy RTTY much better if I used the LOwtones command. What this did was that instead of using the 2125Hz Mark with the Space shifted above that, it used the European "low_tones" in which the Space is 1275Hz and then the Mark is shifted above that. I think I also had to use the INVert command.

So perhaps you could implement these low tones.

73 de Pete
ve5va.qrp@usask.ca

6. Re: [HFSIG:1167] Re: QPSK BANDWIDTH ?

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 3 Jun 1996 10:27:36 -0800
To: hfsig@tapr.org
Message-ID: <58825D8058E@frl.orst.edu>
Rendered body type: text/plain

Hi Tom,

Thanks for the insight - I am learning a lot. Seems like I'm heading in the right direction.

For those that might be wondering what this is about; I have been exploring some of the materials in Tom's soon-to-be-released book on modem design. The section on pulse shaping is particularly good and includes extensive examples that one can play with using an Excel spreadsheet.

The pulses that need to be shaped are from the QPSK I/Q channels prior to modulation and they are indeed approximately square waves - I chose an alpha factor according to what I see being used in V32 modems - but I'll play with some of the other factor to see the effects. In my case the raised cosine filter order is 65 and the FFT program uses 4096 point FFTs that produces some 2 Hz resolution (I do assume that the program applies windows - it's the VE2IQ program).

Pulse shaping has introduced new problems with my clock and carrier acquisition algorithms, mainly because they are very much based on signal energy over a baud. Pulse shaping cleans the signal up to the extent that a lot more work is now needed to obtain and maintain reliable clock and carrier sync. I probably now need to add a scrambler to whiten the spectrum. What a lot of fun!

--Johan

7. Re: [HFSIG:1168] Center freq for rtty ?

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 3 Jun 1996 10:35:12 -0800
To: hfsig@tapr.org
Message-ID: <58846303B36@frl.orst.edu>
Rendered body type: text/plain

Hi Danie,

The choice of 2125/2295 tone pair goes back a long way. It however, probably is not the best choice and you will find that the European standard specifies 1275/1445.

I dont know of any technical reason, besides IF and IF filters that makes one better than the other, besides some DSP algorithms that works better when you have more cycles of the waveform within a signaling element. You saw a lot of these kinds of technical arguments in the "old days" when folks used cassette tapes to store computer data.

Not sure this helps.

--Johan

8. Re: [HFSIG:1170] Re: QPSK BANDWIDTH ?

From: Clifford Buttschardt <cbuttsch@slonet.org>
Date: Mon, 3 Jun 1996 12:37:44 -0700 (PDT)
To: Johan Forrer <FORRERJ@frl.orst.edu>
Cc: hfsig@tapr.org
Message-ID: <Pine.SOL.3.93.960603122853.14881C-100000@spork.callamer.com>
Rendered body type: text/plain

Hi Johan and Tom. I am not sure from your message who Tom might be! In any case, I think that the use of 2125/2975 hertz for rtty eminated from the first use of AFSK on sidband radios. The idea here was to place the tones as high in frequency as possible to avoid any harmonics from being radiated. The lower tones stand a chance that components will fall in the passband. There also were thoughts that the Q of 88 mHy inductors were larger at the higher frequencies. (that's all we had to work with then!)
    Let me suggest that you check carefully with Bill DeCarle regarding any programs of his supposedly written for windows. It was our intent to keep things as simple as possible and windows is as far from that concept as could be imagined! Cliff Buttschardt W6HDO/K7RR

On Mon, 3 Jun 1996, Johan Forrer wrote:

> Hi Tom,
>
> Thanks for the insight - I am learning a lot. Seems like I'm heading in the
> right direction.
>
> For those that might be wondering what this is about; I have been
> exploring some of the materials in Tom's soon-to-be-released book on
> modem design. The section on pulse shaping is particularly good and
> includes extensive examples that one can play with using an Excel
> spreadsheet.
>
> The pulses that need to be shaped are from the QPSK I/Q channels prior to
> modulation and they are indeed approximately square waves - I chose an alpha
> factor according to what I see being used in V32 modems - but I'll play
> with some of the other factor to see the effects. In my case the raised
> cosine filter order is 65 and the FFT program uses 4096 point FFTs that
> produces some 2 Hz resolution (I do assume that the program applies
> windows - it's the VE2IQ program).
>
> Pulse shaping has introduced new problems with my clock and carrier
> acquisition algorithms, mainly because they are very much based
> on signal energy over a baud. Pulse shaping cleans the signal up to
> the extent that a lot more work is now needed to obtain and
> maintain reliable clock and carrier sync. I probably now need to add a
> scrambler to whiten the spectrum. What a lot of fun!
>
> --Johan
>
>
>
>
>

9. Re: [HFSIG:1170] Re: QPSK BANDWIDTH ?

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 3 Jun 1996 13:12:27 -0800
To: hfsig@tapr.org
Message-ID: <58AE5537E04@frl.orst.edu>
Rendered body type: text/plain

Hi Cliff,

Thanks for the note on the high tones - I also built those 88mH demodulators way back - had a lot of fun!

About "windows" - I trust you mean windows like in Hamming or Blackman-Harris, not Microsoft? I use Bill's program under DOS and I suspect the anti-alias filter a little, otherwise, the program works reasonably well, just don't know whether he actually use a window, i.e., a raised cosine or such prior to the FFT. If not, its not a matter of life and death, just the spectrum is a little smeared due to Gibbs affects. That is what Tom McDermot, N5EG, was hinting at. He has written an excellent book on modems

for the radio communications market, and I have had the pleasure to be a
reviewer - the book will be out soon.

Trust you doing OK? Hows the plans for moving coming along?

73's

--Johan

10. Re: [HFSIG:1165] QPSK BANDWIDTH ?

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 3 Jun 1996 20:17:05 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606040317.UAA00621@servo.qualcomm.com>
Rendered body type: text/plain

Johan,

Are you really testing with random data? From your description and
from your results, it sounds like you're using repeating patterns.

You need to use random data and average your results over a fairly
long interval.

Phil

11. Re: [HFSIG:1168] Center freq for rtty ?

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 3 Jun 1996 20:23:33 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606040323.UAA00633@servo.qualcomm.com>
Rendered body type: text/plain

>Why was such high tones selected for RTTY ? (2125Hz mark, 2295Hz
>space) My hf rig shows much better response at 800 to 1000Hz. Would
>it not be better to implement the channel filters for a RTTY DSP modem
>around say 800Hz ?

I suppose it depends on the radio. You should operate near the center
of whatever passband you have to avoid the phase distortion that is
invariably associated with sharp filters near cutoff.

Phil

12. Re: [HFSIG:1167] Re: QPSK BANDWIDTH ?

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 3 Jun 1996 20:28:26 -0700 (PDT)
To: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)
Cc: hfsig@tapr.org
Message-ID: <199606040328.UAA00648@servo.qualcomm.com>
Rendered body type: text/plain

Tom,

Speaking of root raised cosine filters, would you be willing to design
a few for me for use in my SQPSK modem? I have the code for your book,
but I don't have Excel installed. I have the classic Parkes-McClellan
FORTRAN code, but it doesn't seem to be readily adaptable to an arbitrary
amplitude response.

I've been rewriting my modem code to use an arbitrary length FIR
filter. At the moment I've kept the original 8-sample boxcar shape
that was inherent in my earlier explicit integrate-and-dump
implementation.

I'm using impulse signalling, so the transmit and receive filters can
be the same. Give me a alpha of about 0.5 or so (enough to keep the
energy at 1200 baud well within a typical SSB bandwidth). The baud rate
is 1200, the sampling rate is 9600. Will a 33 tap filter be sufficient?

Phil

13. Re: [HFSIG:1170] Re: QPSK BANDWIDTH ?

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 3 Jun 1996 20:47:16 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606040347.UAA00709@servo.qualcomm.com>
Rendered body type: text/plain

Johan,

You might consider the approach I've taken for symbol timing in my satellite SQPSK (aka OQPSK) modem. Rather than extract symbol timing from the data, I do it in a one-shot fashion at the beginning of every packet with a modified Barker sync sequence. Since this is a packet modem with a bounded packet length, I know that my one-shot estimate should hold as long as the crystal clocks on the two ends are reasonably accurate.

For carrier recovery, I also start with a one-shot estimate taken from a carrier preamble, but because carrier frequency is more uncertain I use the preamble estimate to seed an a posteriori 4-phase Costas loop that updates the frequency from block to block.

The Costas loop used to work on raw symbols, but I just changed my code so that once symbol timing is established, the incoming samples are downconverted to DC, LP filtered, rate converted and complex sampled at 2x the baud rate using a fixed LO downconversion frequency derived from the carrier burst in the preamble.

Then the Costas loop only has to correct for the slow phase rotation ("slow" relative to the data rate) caused by any residual error in the preamble frequency estimation (plus any doppler). Although complex samples take more operations to process, I figure I probably save overall from the rate conversion from 9600 samples/sec to 2400. And I've already LP filtered, so I don't have to do that again.

I only need 2x sampling because of the I/Q symbol staggering; true QPSK would only require 1x sampling once symbol timing is established.

Phil

14. Re: [HFSIG:1171] Re: Center freq for rtty ?

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 3 Jun 1996 21:03:34 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606040403.VAA00740@servo.qualcomm.com>
Rendered body type: text/plain

>I dont know of any technical reason, besides IF and IF filters that makes
>one better than the other, besides some DSP algorithms that works better
>when you have more cycles of the waveform within a signaling element. You

I've given this some thought recently, since I noticed some interesting artifacts in my SQPSK modem receive constellation. It's actually inherent in modulating a low frequency carrier with boxcar pulses.

I had chosen a carrier frequency of 1600 Hz and a baud rate of 1200, so there isn't an integral number of carrier cycles per symbol. So when I did a boxcar integrate-and-dump, the I and Q channels aren't perfectly orthogonal even when you have the carrier phase right. I believe this was contributing to about a 1dB implementation loss in my modem.

In the frequency domain, this effect shows up as the lower spectral sidelobes (for unfiltered BPSK) "wrapping around" DC and aliasing back up into the signal spectrum.

I verified my suspicions by testing at a carrier frequency of 2400 Hz, exactly 2x the baud rate. It got rid of these spectral artifacts by overlaying them in "safe" locations (or in the time domain, it ensured an integral number of carrier cycles per symbol integration time.)

But this is not a practical solution for two reasons. First, you can't guarantee that the receiver will get an incoming carrier at exactly 2400 bps, and secondly most SSB radios won't work that high anyway (the first null above the carrier would occur at 3600 Hz, well outside most SSB filters). I chose 1600 Hz in the first place precisely to center the signal spectrum in a typical SSB filter bandwidth.

So I've been rewriting my modem code to include FIR filters in the I and Q channels in both the transmit and receive data paths. With the appropriate filter response, the sidelobes should be low enough to prevent spectral wraparound and aliasing. Or in the time domain, the filter impulse should be small enough at the beginning and end of the symbol time (where the carrier phase transition takes place) to minimize the effect of the fractional carrier cycle in the sampled

filter output.

Of course, another fix would be even "sweeter" -- using two product
detectors in quadrature in the receiver and feeding complex audio
samples to the modem. This would allow the modem to distinguish easily
between positive and negative frequency components. Unfortunately,
most SSB transceivers don't have quadrature audio outputs...

Phil

15. Re: QPSK BANDWIDTH ?

From: "Eric E. Silbaugh" <silbaugh@apci.net>
Date: Mon, 3 Jun 1996 23:28:13 -0500
To: Johan Forrer <forrerj@ucs.orst.edu>
Cc: HFSIG@tapr.org
Message-ID: <199606040428.XAA22147@hilly.apci.net>
Rendered body type: text/plain

Johan,

First let's ignore the differential encoding and raised
cosine filetering. Straight QPSK at T bauds/sec should
have the classic sin(x)/x squared power spectrum with a
main lobe width (null-to-null) of T Hz. What you
consider the 'occupied' bandwidth of this signal depends
on your definition; for now let's use the null-to-null
bandwidth.

Now add raised cosine filtering. The frequency response
of the RC filter (or composite of the RX and TX sqrt RC
filters) will multiply the power spectrum of the original
QPSK waveform to provide the resulting power spectrum.
With a 0.2 rolloff, the ideal raised cosine spectrum will
have a bandwidth of 1.2T Hz. The percent excess bandwidth
is equal to the rolloff factor.

In the real world your filter can only approximate the
ideal RC impulse response. Thus, the spectrum may not
be identically zero beyond 0.6T Hz away from the carrier
frequency. If your filter is a good approximation there
should be little 'splatter' beyond 0.6T Hz.

The previous discussion assumed the data symbols were
uncorrelated. Differential encoding adds some memory;
which means your symbols may not be uncorrelated anymore.
The power spectrum of a digitally modulated signal
depends on the autocorrelation of the data. (see
J. G. Proakis, M. Salehi 'Communication Systems
Engineering,' p.537, Prentice-Hall, 1994) I think the
effects of differential encoding on the spectrum should
be minimal. It certainly should not change the major
features of the spectrum.

Sin(x)/x compensation effects should also be minimal,
as long as your sampling rate is well above the
Nyquist rate. Again, compensation should not change
the major features of the spectrum.

All this is a very long way of saying a bandwidth of
about 1.2T Hz (almost T Hz) should be expected for
your RC filtered DQPSK signal of T baud/sec.

I cannot explain the discrete components you noted.
How do you create the signal you feed into the FFT?
How many baud periods do you use in the FFT'd signal?
You will not get a good estimate of the average
power spectrum from signals only a few baud long.

To estimate the power spectrum with more detail I would
create a signal from several tens or hundreds of random
bits (independent) and FFT this long signal. Yes this
will require a long FFT, but it's just like turning down
the resolution bandwidth of a spectrum analyzer. The
spectral resolution of the FFT is inversely proportional
to the time duration of the signal you transform (longer
times mean finer resolution).

To estimate the average power spectrum you will probably
want to use the same signal and compute the periodogram.

Just average the FFTs of shorter segments of the signal.
The FFT segments can overlap, but should probably be an
integer number of bauds in length. You will be trading
off resolution for averaging. See any good DSP book for
details.

No clue as to the suitability of your choosen rolloff
factor. If it works I guess it's good enough! A factor
of 0.2 seems to be a reasonable compromize from a
bandwidth standpoint. How much amplitude fluctuation
in the time-domain does the filtering add?

Hope this helps, sorry for the bandwidth!

Eric, N2NNP

16. Root raised cosine pulse shaping.

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Mon, 03 Jun 1996 23:30:16 -0800
To: HFSIG@tapr.org
Message-ID: <1.5.4.16.19960604073016.317f6a88@ucs.orst.edu>
Rendered body type: text/plain

Hi Tom,

Well ... After fixing a killer bug, the root raised cosine pulse shaping now
works like it is supposed to. The nicest, tightest constellation points that
I have seen thus far and no problem with clock or carrier extraction either.
Very nice eye patterns - I still have to implement pulse shaping filters in
the receiver to gain all the good benefits due to bandwidth reduction. I am
getting excellent results with a rolloff factor of 0.6.

Thanks for all the good hints.

--Johan

17. [HFSIG:1171] Re: Center freq for rtty ? -Reply

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Tue, 04 Jun 1996 20:08:03 +0200
To: hfsig@tapr.org
Message-ID: <s1b3f0c0.001@KIDD.CO.ZA>
Rendered body type: text/plain

...besides some DSP algorithms that works better when you have more
cycles of the waveform within a signaling element.....

I see. Well I will try to implement both and see if there are any practical
differences. I noted that Dave W3HCF is also using the high tones in his
DSP-93 implementation. Dave any comments from your side ?

73 Danie zs6awk

18. Re: [HFSIG:1179] Re: QPSK BANDWIDTH ?

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 3 Jun 1996 23:32:54 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606040632.XAA01122@servo.qualcomm.com>
Rendered body type: text/plain

Eric,

An excellent writeup. One nit:

>cosine filetering. Straight QPSK at T bauds/sec should
>have the classic sin(x)/x squared power spectrum with a
>main lobe width (null-to-null) of T Hz. What you

Shouldn't that be 2T Hz from null to null? E.g., a 1200 baud (2400
bps) QPSK signal with carrier at 1600 Hz would have first nulls at 400
Hz and 2800 Hz.

Those damn factors of 2 are so easy to get wrong, especially when some
of the textbooks talk about bits per sec and others talk about baud (2
bits/sym for QPSK)...

Phil

19. Re: [HFSIG:1182] Re: QPSK BANDWIDTH ?

From: Jim Sanford <JSANFORD@INFI.NET>
Date: Tue, 04 Jun 1996 07:02:15 -0400
To: hfsig@tapr.org
Message-ID: <199606041101.HAA13762@mh004.infi.net>
Rendered body type: text/plain

Phil:
Have enjoyed following all the threads you're into . . .

A couple of points:
        1. For what platform are you doing the OQPSK
satellite modem? I have 2 dsp-12's and would love to
beta test . . .
        2. You made a comment about quadrature outputs
from SSB receivers being nice but not available. I believe
they will be, SOON. I'm working on a design using
Analog Devices DDS's, Mini circuits DBM's, and Maxim's
quadrature modulators/demodulators. This thing will initally
be for HF, but easily mixed/redesigned for vhf/uhf.
The plummeting cost of the silicon is making this cheaper
by the day.

Intend to initally send I and Q to a "stereo" sound card for
easy playing with the DSP, but baseband bandwidth on the
Maxim demods is large enough that the design will not be
limited to audio. Will gladly share details with you, when
the prototype is done. Believe cost will be $100 or so for
the whole works.

This business is getting terribly exciting -- believe we are on
the threshold of some great capabilities at significantly lower
prices than Icom, et al, would have us believe. Thanks for all
you've done for the state-of-the-art.

73, Jim
wb4gcs@amsat.org

>
>Those damn factors of 2 are so easy to get wrong, especially when some
>of the textbooks talk about bits per sec and others talk about baud (2
>bits/sym for QPSK)...
>
        AMEN!!!!!!!

>Phil
>
>
>

20. RE: [HFSIG:1168] Center freq for rtty ?

From: Stephan Sykes <ssykes@ns2.emirates.net.ae>
Date: Tue, 4 Jun 1996 16:25:54 +-400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB5233.827F2140@csb059.emirates.net.ae>
Rendered body type: text/plain

I was told that the reason for the high tones was to place the harmonics of
the tone outside of the IF filter bandwidth. Improved filters and DSP make
this unnecessary now.

Steve Sykes
KD2OM/A61AA

From: Danie Brynard[SMTP:BRYD@kidd.co.za]
Sent: Monday, June 03, 1996 12:30 PM
To: hfsig@tapr.org
Subject: [HFSIG:1168] Center freq for rtty ?

By the way...

Why was such high tones selected for RTTY ? (2125Hz mark, 2295Hz
space) My hf rig shows much better response at 800 to 1000Hz. Would
it not be better to implement the channel filters for a RTTY DSP modem
around say 800Hz ?

I am busy with a rtty dsp modem for the evm and is now wondering
about some of my earlier choices :-) From a DSP point of view would it

not be better to work around 800Hz ? I have used my PK232 with great
success in the past using my 500Hz CW filter on RTTY.

danie zs6awk


```
begin 600 WINMAIL.DAT
M>)\^(@@,`0:0" `$`````````!``$``0>0!@`(````Y 0```````#H``$-@ 0`
M`@````(``@`!!) &`$ #```"````% ````,``# %`````"P`/#@`````"`?\/
M`0```#L``````````@2L?I+ZC$!F=;@#=`0]4`@`````!H9G-I9T!T87!R+F]R
M9P!33510`&AF<VEG0'1A<'(N;W)G```>``(P`0````4```!33510`````!X`
M`S !````#P```&AF<VEG0'1A<'(N;W)G```>`!H``0```!0```!215!/4E0N
M25!-+DY/5$4N3D12`$ `,@"@$]'L$%*[`0,`! P``````P`%#/____\#`!4,
M````$ ,`_@\&````'@`!$ $```!$````3F\@=')A;G-P;W)T('!R;W9I9&5R
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M@2L?I+ZC$!F=;@#=`0]4`@````!H9G-I9T!T87!R+F]R9P!33510`&AF<VEG
M0'1A<'(N;W)G```>``(P`0````4```!33510`````!X``S !````#P```&AF
M<VEG0'1A<'(N;W)G```>`!H``0````@```!)4$TN3D]410,`%0P!`````P#^
M#P8````>``$P`0```!$````G:&9S:6= =&%P<BYO<F<G``````(!"S !````
M% ```%--5% Z2$9324= 5$%04BY/4D<``P``.0``````"`10Z`0```! ```!F
M!!_Z2KW/$;"Y1$535 ``"P! .@$````"`?8/`0````0````````&=:(!"( '
M`!@```!)4$TN36EC<F]S;V9T($UA:6PN3F]T90`Q" $$@ $`* ```%)%.B!;
M2$9324<Z,3$V.%T@0V5N=&5R(&9R97$@9F]R(')T='D@/P`L# $%@ ,`#@``
M`,P'!@`$`! `&0`V``(`/@$!!@`'`!@```!)4$TN36EC<F]S;V9T($UA:6PN
M3F]T90`Q" $@@ ,`#@```,P'!@`$``8`$@`)``(```$!"8 !`"$````V-C T
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M`",````````,`)@``````"P`I``$````#`#8``````$ `.0! X8WY$%*[`1X`
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M='D@/P`"`7$``0```!8````!NU&\F=_Z'P1GO4H1S["Y1$535 `````>`!X,
M`0````4```!33510`````!X``PP!````%P````-S>6ME<T!E;6ER871E<RYN
M970N864```,`!A DJK%B`P`'$*\"```>``@0`0```&4```!)5T%35$],1%1(
M05142$5214%33TY&3U)42$5(24=(5$].15-705-43U!,04-%5$A%2$%234].
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M``````(!"1 !````J0,``*4#```X!@``3%I&=>RL$<G_``H!#P(5`J@%ZP*#
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M( M18QP0'.,*P 1@Y0,`8P0@;V8;XQUR'W"<=70`D VP'W9)1AR05P,0$] %
MP&(`<&0#\&0I&Z N(!K@;1-0;W8/"8 A500@(>$@1%-0&B `P&L>@@0`('5N
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M)<P*]"!L:3$X, +1:2UX,30T#? ,T"I3"UDQKC8*H -@$]!C!4 M+'>O"H<K
M*PPP*_9&`V$Z+7Z?*_8,@B/0`' (D"!")5 ";@L16U--5% Z0$)2641 :R"
M9,(N!: N>F%=+1\N+2\&8 (P+U\P:TT"(&1A\'DL($HDP!P0&4 WX! Q.3DV
M.( R.C,P,"!033-O+BU4;]LUKS!K: /0'3! `9 34!HN!;!G.6\T?G5B:@<L
M,3N/,&M;2$9320A'.C$KP#A=($-G-7$AH0-097$<DP`@=,TE8#\H?RF#,S8J
M]QI%G2OV0B5@&_(;$'DN2&"9)<Q7:"5@&Q)S=1%P.QT:$;!L+#$B\@6Q4E1$
M5%E$H" H,CC@-21(>B01<FLWX#(RMCE,<0J%<PJP'G I(G"^325@/6 <(!TP
M2>!H)9#U!"!M2@)B$< ADA803>#_`B 1L".0!4 IX1X"&3 I\+M,@")@5PA@
M&W *A6D%0/\E@ 5 3[!/IAX1!W +4!/@?S5Q&^,1<231`R C)ARB8?=+M"/C
M!'%M"H4*P A@([$_)3 E8"GA3(%$O1KP86WY(<!U<TF14I =4%7P1&/^9$W@
M5I0<ER; 69 CHB21_R6!&P`WD09Q&1!6]@;@(%"W2> #<""C;25@'$!R*;";
M(:$1<&\?0 >1.BU.,'\O$E7A(^)04 N !4 ?@7;_")!<LE'Q4H$*A5+/7-!,
MT/]75E@5&N$1@";16< B\5ZQ\%!+,C,1X%H#"<$;P/]-=DH`)0)2@ .@&_(*
ML!/ EV5173%>HC58(T-7(58_''%+PB)@)<PWL#%2>G/P-F%W:R7,13]&3RP%
M"PJ%%3$`<' ````#`! 0``````,`$1 `````0 `', #JTQ*\4;L!0 `(, #J
MTQ*\4;L!`@$4.@$````0````9@@?^DJ]SQ&PN41%4U0``!X`/0`!````!0``
+`%)%.B ``````(P`
`
end
```


21. Re: [HFSIG:1156] Re: DSP for Linux (was: Modem audio sample)

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Tue, 4 Jun 1996 09:43:15 -0400
To: hfsig@tapr.org
Message-ID: <1B43D890.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

Johan,

> I would interested in the URLs you are speaking about. Also, what is
> the "size" of the box you are refering to?

Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1156] Re: DSP for Linux (was: Modem audio sample)
Author: hfsig@tapr.org at BALT.SMTP
Date: 5/23/96 9:02 PM

Hi Steve,

Another gem for Linux is Matcom; its a Matlab to C++ translator. Then there is Octave, which is a Matlab look-alike. Of course this software will only mean something to you if you are familiar with Matlab. Matlab allows for pretty sophisticated DSP evaluation and simulation. Also don't forget about Ptolemy - really nice but you need a fairly big Linux box to use it.

I don't have the URL's available at the moment, but can let you know if there is interest.

--Johan

At 01:55 PM 5/23/96 -0500, you wrote:
>At 07:22 AM 5/23/96 -0500, you wrote:
>> Steve,
>>
>> Where do you find this great stuff???
>
>The best way to keep up with what's new with Linux is to read the
>comp.os.linux.announce newsgroup. There's also a web site that archives
>them, the URL is http://www.cs.helsinki.fi/%7Ewirzeniu/linux/cola.html.
>
>Otherwise, I don't watch a lot of TV. I'd much rather read and experiment :-).
>
>73,
>
> - Steve, N7HPR
>
>srbible@gnatnet.net
>n7hpr@amsat.org
>n7hpr@tapr.org
>
>

22. Re: [HFSIG:1177] Re: QPSK BANDWIDTH ?

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Tue, 4 Jun 1996 08:36:30 -0800
To: hfsig@tapr.org
Message-ID: <59E4CAB540F@frl.orst.edu>
Rendered body type: text/plain

Tom, Phil and Eric,

Much thanks for the excellent ideas and feedback.

I do suspect that some of the spectral lines I'm seeing with my QPSK signal is due to the nature of the cyclic patters that is being sent. I'll work on a PRBS generator as suggested.

Clock and carrier extraction is a fascinating subject and I'll certainly look at Phil's ideas. For this HF modem, the symbol rate is so low that FFT's and all sort of things alike may be put to good use, and I'll get to that shortly. The main thing though is that HF is not too friendly to true synchronous detection methods - the key to success is to go for robustness in algorithm design. I think I have some ideas based on robustness (in the statistical sense) that seem work reasonably well, but need to be put through the wringer.

I have been looking at epoch synch methods but was seeing enough drift even with two similar sound cards over a few seconds of signal. However, over a frame duration of a second or two, I suspect epoch sync may just work out fine.

Keep up the good work.

--Johan

23. Re: [HFSIG:1183] Re: QPSK BANDWIDTH ?

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Tue, 4 Jun 1996 08:50:37 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960604084018.15342A-100000@ucs.orst.edu>
Rendered body type: text/plain

Hi Jim,

Sounds interesting. Would you care to elaborate more?

I am sort of midway in a project building a VHF/UHF digital receiver for satellite work. It uses a CATV tuner for the RF front-end which has a 45MHz IF, 20 MHz A/D fed into a Harris DDC that feeds I/Q outputs into a stereo CODEC connected to a EZ-kit (ADSP-2181). I use an AD7008 DSS and a PLL for frequency control.

I thought I would stay away from HF for a while until I learn a bit more about the RF digital dynamic range, AGC, and it's affects. Guess there is a lot to learn.

On Tue, 4 Jun 1996, Jim Sanford wrote:

> Phil:
> Have enjoyed following all the threads you're into . . .
>
> A couple of points:
> 1. For what platform are you doing the OQPSK
> satellite modem? I have 2 dsp-12's and would love to
> beta test . . .
> 2. You made a comment about quadrature outputs
> from SSB receivers being nice but not available. I believe
> they will be, SOON. I'm working on a design using
> Analog Devices DDS's, Mini circuits DBM's, and Maxim's
> quadrature modulators/demodulators. This thing will initally
> be for HF, but easily mixed/redesigned for vhf/uhf.
> The plummeting cost of the silicon is making this cheaper
> by the day.

I would be interested to see what sideband rejection youre going to get. Don't be dissapointed if it's only about 40 dB.

> > Intend to initally send I and Q to a "stereo" sound card for
> easy playing with the DSP, but baseband bandwidth on the
> Maxim demods is large enough that the design will not be
> limited to audio. Will gladly share details with you, when
> the prototype is done. Believe cost will be $100 or so for
> the whole works.
>
> This business is getting terribly exciting -- believe we are on
> the threshold of some great capabilities at significantly lower
> prices than Icom, et al, would have us believe. Thanks for al

--Johan, KC7WW

24. Tom's books

From: "Alberto di Bene (xx39-2-596.25744)" <dibene@VNET.IBM.COM>
Date: Tue, 4 Jun 96 17:55:39 EDT
To: hfsig@tapr.org
Message-ID: <199606041558.KAA05916@tapr.org>
Rendered body type: text/plain

> For those that might be wondering what this is about; I have been
> exploring some of the materials in Tom's soon-to-be-released book on
> modem design. The section on pulse shaping is particularly good and

 Now you got me curious... any chance to have the title and ISBN of that
 book ? Thanks.

 73 de
 Alberto di Bene, I2PHD

25. Re: [HFSIG:1184] RE: Center freq for rtty

From: Clifford Buttschardt <cbuttsch@slonet.org>
Date: Tue, 4 Jun 1996 10:33:42 -0700 (PDT)
To: Stephan Sykes <ssykes@ns2.emirates.net.ae>

Cc: Bill DeCarle <bill@ietc.ca>, hfsig@tapr.org
Message-ID: <Pine.SOL.3.93.960604102455.17919D-100000@spork.callamer.com>
Rendered body type: text/plain

HI All. I was one of probably many that indicated that the high tones were used to keep harmonics down when using AFSK in SSB rig. Indeed, nowdays there is no reason to use such high tones and commercially, those higher tones are obsolete! We used such high frequencies as we simply duplicated telephone channel filters in which 3 KHz was the upper limit. Not a one of us knew enough about fileter design to change the system!
     I fully concer with the use of 800 Hz as a center frequency. All of Bill DeCarles work is centered there as is all my filter designs. It seems a good selection although a short time ago I was thumping for 500 Hertz as it would have make synthesizer design simpler...I lost!!
    Danie- I have no way of knowing what you said in the Winmail portion of your note. Can't cope with that! sorry Cliff Buttschardt W6HDO/K7RR

On Tue, 4 Jun 1996, Stephan Sykes wrote:

> I was told that the reason for the high tones was to place the harmonics of
> the tone outside of the IF filter bandwidth. Improved filters and DSP make
> this unnecessary now.
>
> Steve Sykes
> KD2OM/A61AA
>
> ----------
> From: Danie Brynard[SMTP:BRYD@kidd.co.za]
> Sent: Monday, June 03, 1996 12:30 PM
> To: hfsig@tapr.org
> Subject: [HFSIG:1168] Center freq for rtty ?
>
> By the way...
>
> Why was such high tones selected for RTTY ? (2125Hz mark, 2295Hz
> space) My hf rig shows much better response at 800 to 1000Hz. Would
> it not be better to implement the channel filters for a RTTY DSP modem
> around say 800Hz ?
>
> I am busy with a rtty dsp modem for the evm and is now wondering
> about some of my earlier choices :-) From a DSP point of view would it
> not be better to work around 800Hz ? I have used my PK232 with great
> success in the past using my 500Hz CW filter on RTTY.
>
> danie zs6awk
>
>
>
>
> begin 600 WINMAIL.DAT
> M>)\^(@@,`0:0" `$```````!``$``0>0!@`(````Y 0```````#H``$-@ 0`
> M`@````(``@`!!) &`$ #```"````% ````,``# %````"P`/#@`````"`?\/
> M`0```#L``````````@2L?I+ZC$!F=;@#=`0]4`@````!H9G-I9T!T87!R+F]R
> M9P!33510`&AF<VEG0'1A<'(N;W)G```>``(P`0````4```!33510`````!X`
> M`S !````#P```&AF<VEG0'1A<'(N;W)G```>`!H``0```!0```!215!/4E0N
> M25!-+DY/5$4N3D12`$ `,@"@$]'L$%*[`0,`! P``````P`%#/____\#`!4,
> M````$,`_@\&````'@`!$ $```!$````3F\@=')A;G-P;W)T('!R;W9I9&5R
> M('=A<R!A=F%I;&%B;&4@9F]R(&1E;&EV97)Y('1O('1H:7,@<F5C:7!I96YT
> M+@`>``$P`0```!$````G:&9S:6= =&%P<BYO<F<G``````(!"S !````% ``
> M`%--5% Z2$9324= 5$%04BY/4D<``P``.0`````"`1(Z`0```#L``````````
> M@2L?I+ZC$!F=;@#=`0]4`@````!H9G-I9T!T87!R+F]R9P!33510`&AF<VEG
> M0'1A<'(N;W)G```>`!,Z`0```!$````G:&9S:6= =&%P<BYO<F<G``````(!
> M%#H!````% ```%--5% Z2$9324= 5$%04BY/4D<`"P! .@$````"`?8/`0``
> M``0`````````%#@````,``# &````"P`/#@$````"`?\/`0```#L``````````
> M@2L?I+ZC$!F=;@#=`0]4`@````!H9G-I9T!T87!R+F]R9P!33510`&AF<VEG
> M0'1A<'(N;W)G```>``(P`0```4```!33510`````!X``S !````#P```&AF
> M<VEG0'1A<'(N;W)G```>`!H``0````@```!)4$TN3D]410,`%0P!`````P#^
> M#P8````>``$P`0```!$````G:&9S:6= =&%P<BYO<F<G``````(!"S !`````
> M% ```%--5% Z2$9324= 5$%04BY/4D<``P``.0`````"`10Z`0```! ```!F
> M!!_Z2KW/$;"Y1$535 ``"P! .@$````"`?8/`0````0````````&=:(!"('
> M`!@```!)4$TN36EC<F]S;V9T($UA:6PN3F]T90`Q" $$@ $`* ```%)%.B!;
> M2$9324<Z,3$V.%T@0V5N=&5R(&9R97$@9F]R(')T='D@/P`L# $%@ ,`#@``
> M`,P'!@`$`! `&0`V``(`/@$!!@`''!@```!)4$TN36EC<F]S;V9T($UA:6PN
> M3F]T90`Q" $@@ ,`#@```,P'!@`$``8`$@`)``(```$!"8 !`"$````V-C T
> M,49&031!0D1#1C$Q0C!".30T-#4U,S4T,# P, `#!P$#D 8`5 4``!,`````+
> M`",````````,`)@``````"P`I``$````#`#8``````$ `.0! X8WY$%*[`1X`
> M< `!````* ```%)%.B!;2$9324<Z,3$V.%T@0V5N=&5R(&9R97$@9F]R(')T
> M='D@/P`"`7$``0```!8````!NU&\F=_Z'P1GO4H1S["Y1$535 `````>`!X,
> M`0````4```!33510`````!X`'PP!````%P```'-S>6ME<T!E;6ER871E<RYN
> M970N864```,`!A DJK%B`P`'$*\"````>``@0`0```&4```!)5T%35$],1%1(

> M05142$5214%33TY&3U)42$5(24=(5$].15-705-43U!,04-%5$A%2$%234].
> M24-33T942$543TY%3U544TE$14]&5$A%249&24Q415)"04Y$5TE5A)35!2
> M``````(!"1 !````J0,``*4#```X!@``3%I&=>RL$<G_``H!#P(5`J@%ZP*#
> M`% "\@D"`&-H"L!S970R-P8`!L,"@S(#Q0(`<')"<1'B<W1E;0*#,W<"Y <3
> M`H!]"H (SPG9._$6#S(U-0* "H$-L0M@P&YG,3 S.0K[%%$%"_)C`$ @22!W
> M8?4$('0&\&0;0!& !4 ;H-QE(!80&R "(" "$ 7 H1OR:&EG:!M!;@>1]1L4
> M(M18QP0'.,*P 1@Y0,`8P0@;V8;XQUR'W"<=70`D VP'W9)1AR05P,0$] %
> MP&(`<&0#\&0I&Z N(!K@;1-0;W8/"8 A500@(>$@1%-0&B `P&L>@@0`('5N
> M#QV0'G $$ K >2!N;^QW+@J%"H53$] BX!'Q!21 <PJ%2T0R3TW +T$V,4%!
> M)<P*]"!L:3$X, +1:2UX,30T#? ,T"I3"UDQKC8*H -@$]!C!4 M+'>O"H<K
> M*PPP*_9&`V$Z+7Z?*_8,@B/0`' (D"!")5 ";@L16U--5% Z0$)2641 :R"
> M9,(N!: N>F%=+1\N+2\&8 (P+U\P:TT"(&1A\'DL($HDP!P0&4 WX! Q.3DV
> M.(R.C,P,"!033-O+BU4;]LUKS!K: /0'3! `9 34!HN!;!G.6\T?G5B:@<L
> M,3N/,&M;2$9320A'.C$KP#A=($-G-7$AH0-097$<DP`@=,TE8#\H?RF#,S8J
> M]QI%G2OV0B5@&_(;$'DN2&"9)<Q7:"5@&Q)S=1%P.QT:$;!L+#$B\@6Q4E1$
> M5%E$H" H,CC@-21(>B01<FLWX#(RMCE,<0J%<PJP'G I(G"^325@/6 <(!TP
> M2>!H)9#U!"!M2@)B$< ADA803>#_`B 1L".0!4 IX1X"&3 I\+M,@")@5PA@
> M&W *A6D%0/\E@ 5 3[!/IAX1!W +4!/@?S5Q&^,1<231`R C)ARB8?=+M"/C
> M!'%M"H4*P A@([$_)3 E8"GA3(%$O1KP86WY(<!U<TF14I =4%7P1&/^9$W@
> M5I0<ER; 69 CHB21_R6!&P`WD09Q&1!6]@;@(%"W2> #<""C;25@'$!R*;";
> M(:$1<&\?0 >1.BU.,'\O$E7A(^)04 N !4 ?@7;_")!<LE'Q4H$*A5+/7-!,
> MT/]75E@5&N$1@";16< B\5ZQ\%!+,C,1X%H#"<$;P/]-=DH`)0)2@ .@&_(*
> ML!/ EV5173%>HC58(T-7(58_''%+PB)@)<PWL#%2>G/P-F%W:R7,13]&3RP%
> M"PJ%%3$`<' ````#`! 0``````,`$1 `````0 `', #JTQ*\4;L!0 `(, #J
> MTQ*\4;L!`@$4.@$`````0````9@0?^DJ]SQ&PN41%4U0``!X`/0`!````!0``
> +`%)%.B ``````(P`
> `
> end
>
>

26. Re: [HFSIG:1183] Re: QPSK BANDWIDTH ?

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 4 Jun 1996 13:07:12 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606042007.NAA10211@servo.qualcomm.com>
Rendered body type: text/plain

>   1. For what platform are you doing the OQPSK
>satellite modem? I have 2 dsp-12's and would love to
>beta test . . .

The platform is a generic 486 or Pentium PC running my KA9Q NOS TCP/IP package. The modem will be integrated in as an interface driver for a SoundBlaster 16 card *without* any special DSP chips.

Your work with the 'digital friendly' receiver sounds very interesting! I do hope that whatever you come up with can be easily replicated by the average amateur -- in my experience, that's where many promising amateur hardware projects fall down. It's the main reason I've become so interested in using mass-market PC hardware to the greatest possible extent, doing everything else in software. But there are definite limits to the available hardware (particularly in the radios) so your work is most welcome. I agree, this stuff ought not to be as expensive and complicated as it seems to be.

At Dayton I looked at some of the companies selling after-market crystal filters for commercial SSB transceivers with an eye toward modifying them for wider baseband bandwidths. Ideally I'd like a ~20 KHz bandwidth so I could fully utilize the A/D bandwidth of a PC sound card. I could even create a new "IF" at, say, 10 KHz and do my final "product detection" in DSP software.

The guy behind the Fox-Tango table, being used to selling very sharp and narrow crystal filters for HF contest and DX use, couldn't figure out why in the world I would want to make my rig go *wider*. Ah, the "narrowband mindset" seems pretty well entrenched. :-)

Phil

27. RE: [HFSIG:1189] RE: Center freq for rtty

From: LAITINEN@ESHOP.UOREGON.EDU
Date: Sun, 4 Jun 1995 13:48:40 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <950604134841.b9b@ESHOP.UOREGON.EDU>
Rendered body type: text/plain

Gee, I thought that the use of 2125/2975 Hz for AFSK tones in the old days was related to running AFSK RATT on 2-meters "AM" (and then

eventually FM). Tuning the AFSK generator and TU filters to the proper
frequencies was done using standard tuning forks and lisajous (spelling?)
patterns on an oscilloscope. Standard landline VFCT (Voice Freq Carrier
Telegraph) systems (as I recall) were set up with 85-Hz channels and the
tools (i.e., tuning forks) were around for tuning such things... Thus the
harmonic relationships of the old 2125/2975 tones to these tuning fork
standards were important...

My first 2-meter RATT setup used an ARC-5/T-23 surplus aircraft radio
modulated by an Eico 730K modulator. The RATT gear was a Model 19 with a
W2JAV TU (the one with four neon bulbs on the front).... The receiver was
the 417A converter that was popular in the early 1960's for Project OSCAR...

Larry, WA6JYJ/7

28. Re: [HFSIG:1188] Tom's books

From: Greg Jones <wd5ivd@tapr.org>
Date: Tue, 4 Jun 1996 20:49:22 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199606050149.UAA29453@tapr.org>
Rendered body type: text/plain

Title: Wireless Digital Communications: Design and Theory
By: Tom McDermott, N5EG
ISBN: 0-9644707-2-1

Anticipated price will be $39.00

With luck it will be available from TAPR the end of August/first of Sept.
Tom is currently doing small technical corrections and the proof-reader is
working over grammer issues. I get those elemtns and then with luck we go
to thew e printers.

I think there is a web page already on the tapr web server under
publications. If it isn't there -- then I have it ready, but haven't put it
up yet,. Has a complete index...that is if it is up.

This has been a year and a half project. Will be glad to get it finsihed.

Cheers - Greg, WD5IVD

>
> > For those that might be wondering what this is about; I have been
> > exploring some of the materials in Tom's soon-to-be-released book on
> > modem design. The section on pulse shaping is particularly good and
>
> Now you got me curious... any chance to have the title and ISBN of that
> book ? Thanks.
>
> 73 de
> Alberto di Bene, I2PHD
>
>

29. [HFSIG:1191] RE: Center freq for rtty -Reply

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Wed, 05 Jun 1996 19:56:05 +0200
To: hfsig@tapr.org
Message-ID: <s1b53ec4.049@KIDD.CO.ZA>
Rendered body type: text/plain

Larry now that is interesting. Thanks for the bit of history.

danie

30. Sine wave PROM

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Wed, 5 Jun 1996 09:48:54 -0400
To: hfsig@tapr.org
Message-ID: <1B5ADB10.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

I am trying to locate information on generating sine wave data for a
     EEPROM. I had an article (Radio Electronics) describing this by using
     a C program to generate the data. However, I can't find the article.

     Can someone help with this problem?

73s de
Tony, KE4ATO

31. Re: QPSK BANDWIDTH ?

From: "Eric E. Silbaugh" <silbaugh@apci.net>
Date: Wed, 5 Jun 1996 22:40:02 -0500
To: hfsig@tapr.org
Message-ID: <199606060340.WAA07740@hilly.apci.net>
Rendered body type: text/plain

Phil,

Good nit! Fourier is probably turning in his grave.

I was visualizing the spectra at baseband, wrote about it as
RF modulated, and forgot to multiply the bandwidths by two.
Arrrgggh! Thus, the first null in a QPSK signal of T baud will
be T Hz away from the carrier; the main lobe width is 2T Hz;
and the bandwidth with raised cosine filtering, rolloff factor
of 0.2, will be 2.4T Hz.

Did I get it right this time?

Eric, N2NNP

32. Further thoughts on pulse shaping for HF

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Thu, 6 Jun 1996 11:20:01 -0800
To: hfsig@tapr.org
Message-ID: <5D1078A41EC@frl.orst.edu>
Rendered body type: text/plain

Hi folks,

Another thought/question on pulse shaping for HF;

Consider the recovered signals from a QPSK modem where the final product
is a raised cosine - I do encounter a little annoying sideffect. In the
vacinity of the sampling point, one has to read both I and Q - for QPSK
case, one expects that either I or Q must be near zero, i.e. a pair from
the set {1,0 -1,0 0,1 0,-1 for I/Q}.

Unfortunately, however, the pulse shaping procedure forces part of the pulse
beyond the +/-T mark, to go negative, which, combined with a bit of jitter
introduced by the channel effects, causes the channel that is supposed to
be zero, to an off-zero value. These uncertainties are further aggravated
that this error in the zero value, changes fairly rapidly in the vacinity of
the sampling point. It would have been nice if things around the zero
transition were a bit more stable.

Now, if I were to use another shape function like Dolph-Chebychef, or
the Blackman-Harris shape for minimum sidelobes, instead of raised cosines,
I would not have true nyquist pulses - the tradeoff is a some of ISI,
however, would'nt the result be more robust against this jitter problem as
described above. I.e., would'nt the tradeoff between ML detection and
robustness for clock/carrier extraction, which is more crucial, make
sense?

Comments or suggestions would be appreciated.

--Johan, KC7WW

33. Re: [HFSIG:1195] Re: QPSK BANDWIDTH ?

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 6 Jun 1996 18:18:57 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606070118.SAA15345@servo.qualcomm.com>
Rendered body type: text/plain

>I was visualizing the spectra at baseband, wrote about it as
>RF modulated, and forgot to multiply the bandwidths by two.
>Arrrgggh! Thus, the first null in a QPSK signal of T baud will
>be T Hz away from the carrier; the main lobe width is 2T Hz;
>and the bandwidth with raised cosine filtering, rolloff factor
>of 0.2, will be 2.4T Hz.

>Did I get it right this time?

Almost. The null-to-null bandwidth of 2T Hz looks right, but with
raised cosine filtering the bandwidth will be *less* than the
unfiltered null-to-null bandwidth.

Phil

34. Re: [HFSIG:1196] Further thoughts on pulse shaping for HF

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 6 Jun 1996 18:31:38 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606070131.SAA15427@servo.qualcomm.com>
Rendered body type: text/plain

>Consider the recovered signals from a QPSK modem where the final product
>is a raised cosine - I do encounter a little annoying sideffect. In the
>vacinity of the sampling point, one has to read both I and Q - for QPSK
>case, one expects that either I or Q must be near zero, i.e. a pair from
>the set {1,0 -1,0 0,1 0,-1 for I/Q}.

Well, you can use any convention you like, but the usual one is to
signal with the set { (a,a), (-a,a), (-a,-a), (a,-a) } with a=sqrt(2)
times the signal magnitude and with the detectors looking
independently at the I and Q components. That is, since I and Q are
(ideally) orthogonal the I detector simply produces 1 or -1 (or a soft
decision sample somewhere in this range) without caring what is going
on in the Q channel. And vice versa.

This particular constellation is used by the 4-phase costas loop,
which will drive the I and Q components to equal amplitude. You can of
course use the constellation you described, but then you have to
rotate your decision regions so they're no longer aligned with the I
and Q axes.

Phil

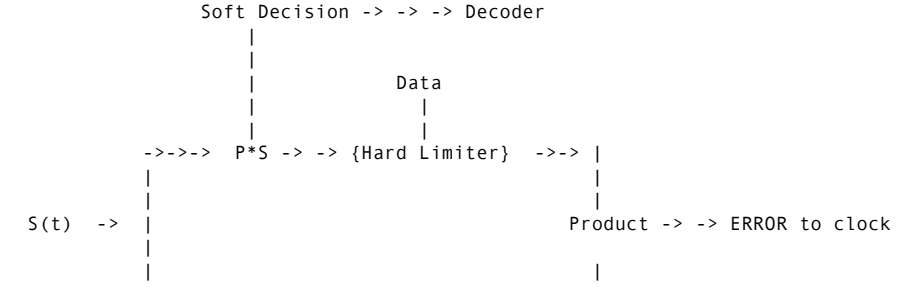
35. Re: [HFSIG:1196] Further thoughts on pulse shaping for HF

From: Bob McGwier <n4hy@ccr-p.ida.org>
Date: Fri, 7 Jun 1996 10:57:40 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199606071457.KAA00644@growler.ccr-p.ida.org>
Rendered body type: text/plain

A good approximation to the ML detector for clock AND data together is what
you need Johan. Take the final pulse shape (sampled) and its derivative
(Guess at this or approximate it. If it is raised cosine, you may compute
it). Call these signals P(i) and P'(i) , i = -n, -(n-1), ..., 0,1,2,...
(n-1),n respectively. Suppose for this exercise that you have an approximate
time for the CENTER (the sampling time) for the I and Q channel bauds.

At the putative sampling time t, compute P*S(t) and P'*S(t). That is,
convolve the pulse shape, and its derive with the baseband signal S and do
it for S from the I and Q channel. The SIGN of the P*S is the guessed at
symbol and SIGN(P*S(t))*tanh [P'*S(t)] = ERROR signal to be multiplied
by the gain which will determine your "loop bandwidth". If you are running
a first order loop for clock, this signal (gain*ERROR) is applied to the
phase of your clock and then you march along to the next symbol time. This
is MUCH more robust, stable, etc, than using clock edges, or zero crossings,
and so long as you have the pulse shape "about right", even ISI, jitter,
does not disturb you much. This was proposed a long time ago, before the
advent of DSP devices by Umberto Mengali. It was proven by yours truly that
this converges under certain conditions which are near to reality to the
maximum likelihood estimator.

A diagram of this might be more useful to nonmathematicians ;=):



```
          |                                    |
          ->->-> P'*S -> -> -> tanh -> -> -> ->->|
```

The obvious way of combining this for each channel would be to average, or in the case of staggered, run the clock at two times the individual symbol rate and alternate the channels feeding the clock. You may run this as a second order phase locked loop if you believe the clock frequency is also unstable. I used this to derive clock for a really noisy signal from Venus (VEGA Balloon probe). STAY AWAY FROM CLOCK EDGES in a really good modem. Implement the tanh with a table look up and/or some approximation that you find useful. The major condition is that the pulse shape in a idealized channel has zero's at the baud time for neighboring bauds. I have found it to be quite robust. I use Passband equalization whenever possible. The two possible outputs for data depends on whether or not there is error correction coding that needs soft decisions. You will pay larger than a 2 dB penalty to use the hard decision tap for your decoder on HF channels should you decide to go that way. There are other forms of this, but this one is one of the easiest to implement.

Bob


```
Dr. Robert W. McGwier              | n4hy@ccr-p.ida.org: ham radio, scouts,
Center for Communications Research | astronomy, golf (o yea, & math!)
Princeton, N.J. 08520              | Cmte member Troop 5700, ACM Pack 53,
(609)-279-6240(v) (609)-924-3061(f)| Frmr Cncl Comm. GWC 362  Sanhican #2 WWW,
(609)-443-8963 (h)                 | I used to be a Buffalo . . . NE III-120
Explorer Post 995 advisor          | proud parent in Brownie Troop 196
```


36. Re: [HFSIG:1199] Re: Further thoughts on pulse shaping for HF

From: Johan Forrer <forrerj@ucs.orst.edu>

Date: Sat, 08 Jun 1996 11:40:55 -0800
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19960608194055.3f8fec56@ucs.orst.edu>
Rendered body type: text/plain

Hi Bob,

Thanks for sharing your ideas - also welcome to HFSIG.

At 10:00 AM 6/7/96 -0500, you wrote:
>
>
>A good approximation to the ML detector for clock AND data together is what
>you need Johan.

I suspect you are right. The question that have been on my mind lately regards how much of the ideas that we commonly use for synchronous demodulation is worth implementing on HF. As an example, I have been playing with a low baudrate QPSK demodulator over the last few weeks where most of the clock and carrier extraction ideas are based on robust non-syncronous measures. I have had some encouraging results, but everything is not quite working either. The ideas are based roughly on the following two guidelines:

1) Do clock extraction first because
   a) The middle of the baud is where the signal has most energy and there is
      best phase stability (stay away from the transition zones).
   b) Attempt carrier extraction *only* during this stable period.

2) Decision-directed carrier extraction - preferably not based on PLL's or Costas loops.

On the latter subject; I have played and struggled with various ideas - what I found was that the extended Costas loop (i.e., with I/Q sgn and summming of cross products) works very nicely indeed with waveforms that is not overly distorted. In fact in Frerking's book (Digital Signal Processing in Communications Systems, Marvin Frerking ISBN 0442016166 p.444) makes you aware of the fact that you *not* use the output of a matched filter but a lowpass filter to steer the Costas loop. I have found this to be valid.

This is one reason why I'm quite interested in certain constellations that is easier to deal with than others.

>Take the final pulse shape (sampled) and its derivative
>(Guess at this or approximate it. If it is raised cosine, you may compute
>it). Call these signals P(i) and P'(i) , i = -n, -(n-1), ..., 0,1,2,...
>(n-1),n respectively. Suppose for this exercise that you have an approximate
>time for the CENTER (the sampling time) for the I and Q channel bauds.

>
>At the putative sampling time t, compute   P*S(t) and P'*S(t). That is,
>convolve the pulse shape, and its derive with the baseband signal S and do
>it for S from the I and Q channel. The SIGN of the P*S is the guessed at
>symbol and SIGN(P*S(t))*tanh [P'*S(t)] = ERROR signal to be multiplied
>by the gain which will determine your "loop bandwidth". If you are running
>a first order loop for clock, this signal (gain*ERROR) is applied to the
>phase of your clock and then you march along to the next symbol time. This
>is MUCH more robust, stable, etc, than using clock edges, or zero crossings,
>and so long as you have the pulse shape "about right", even ISI, jitter,
>does not disturb you much. This was proposed a long time ago, before the
>advent of DSP devices by Umberto Mengali. It was proven by yours truly that
>this converges under certain conditions which are near to reality to the
>maximum likelihood estimator.

Ah .. I like this idea and will give it some more thought. Could you be so
kind and give us the full references.

>
>
>
>A diagram of this might be more useful to nonmathematicians ;=):
>
>   Soft Decision -> -> -> Decoder
> |
> |
> | Data
> | |
> | |
>   ->->-> P*S -> -> {Hard Limiter} ->-> |
> | |
> | |
>S(t) -> | Product -> -> ERROR to clock
> |
> | |
> | |
>   ->->-> P'*S -> -> -> tanh -> -> -> ->->|
>
>The obvious way of combining this for each channel would be to average,
>or in the case of staggered, run the clock at two times the individual symbol
>rate and alternate the channels feeding the clock. You may run this as
>a second order phase locked loop if you believe the clock frequency is also
>unstable. I used this to derive clock for a really noisy signal from
>Venus (VEGA Balloon probe). STAY AWAY FROM CLOCK EDGES in a really good
>modem.

I guess we agree that the usual synchronous demodulation ideas needs careful
rethinking.

>Implement the tanh with a table look up and/or some approximation
>that you find useful. The major condition is that the pulse shape in
>a idealized channel has zero's at the baud time for neighboring bauds.

Would it thus be an advantage to use such a shaping function; I do know that
the Dolph-chebycheff and Blackman-Harris shapes, for example, was designed
for that.

>I have found it to be quite robust. I use Passband equalization whenever
>possible.

How is this done? using adaptive FIR structures? does this require some kind
of training sequence?

>The two possible outputs for data depends on whether or not there
>is error correction coding that needs soft decisions. You will pay larger
>than a 2 dB penalty to use the hard decision tap for your decoder on HF
>channels should you decide to go that way. There are other forms of this,
>but this one is one of the easiest to implement.
>
>Bob
>
>Dr. Robert W. McGwier | n4hy@ccr-p.ida.org: ham radio, scouts,
>Center for Communications Research | astronomy, golf (o yea, & math!)
>Princeton, N.J. 08520 | Cmte member Troop 5700, ACM Pack 53,
>(609)-279-6240(v) (609)-924-3061(f)| Frmr Cncl Comm. GWC 362 Sanhican #2 WWW,
>(609)-443-8963 (h) | I used to be a Buffalo . . . NE III-120
>Explorer Post 995 advisor | proud parent in Brownie Troop 196
>

>
>
>
>

Pardon if I seem too inquisitive and ask too many dumb questions.

--Johan, KC7WW

37. Stuck at the beginning

From: Pete Hardie <hardie@duke.usask.ca>
Date: Sat, 8 Jun 1996 13:46:57 -0600 (CST)
To: hfsig@tapr.org
Rendered body type: text/plain

In-Reply-To: <1.5.4.16.19960608194055.3f8fec56@ucs.orst.edu>
Message-ID: <Pine.OSF.3.93.960608132158.5705A-100000@duke.usask.ca>
MIME-Version: 1.0
Content-Type: TEXT/PLAIN; charset=US-ASCII

I'm just starting with DSP and thought (after creating some CW FIR
filters) that I'd try to implement a 1200 baud packet TNC on the EZ-KIT
Lite.
I had assumed that implementing the modulation side of this was going to
be easy but I've run into a major hurdle that I can't figure out.

Generating the NRZI data and modulating it as a continuous phase FSK audio
signal was easy. I have set up the DSP program so that it reads KISS
packets from the RS-232 port and then generates the FSK. However, if I put
the audio from the Kit into either my KAM or Tiny-2 neither of them will
decode a packet. I tried this with passall on so that even if the CRC were
bad they should print something out - but they don't. In both cases their
DCD LED does come on while the packet audio is present so they're seeing
something. (BTW - I have used a frequency counter to check the frequency
and stability of the generated tones and of the data bit rate and they're
bang on).

Next step was that I wrote two programs on my Amiga. One samples the
parallel port (I called this program "scope") and the other (called
"nrzi") takes the results from "scope" and decodes it back to the original
data bits (and if the CRC is OK it also attempts to print the AX25
header).

I connect the NRZI digital output of the TNC's modem (3105) into the
parallel port and "scope" samples this at any rate I choose.

If I play the DSP audio into the TNC and sample its modem's NRZI output,
"nrzi" decodes it with no problem. But, if I look at the RS-232 output of
the TNC itself it doesn't send the content of the packet to the computer.

The bit that really throws me is that if I connect the TNC to the audio
from my radio and then run "scope" and "nrzi" on what the TNC hears, the
"nrzi" program decodes "real" packets without any trouble too and this is
using exactly the same sampling rates as I used to sample the audio from
the DSP.

The fact that I can decode the NRZI output from the TNC modem eliminates
several possible problems such as incorrect audio tones and the fact that
the "nrzi" program can decode "real" packets using the modem's NRZI output
suggests that the timing is also correct.

I'm lost. I know this is "back to basics" for most of you but has anyone
any idea where I can start looking for the problem?

73 de Pete
ve5va.qrp@usask.ca

38. Re: [HFSIG:1201] Stuck at the beginning

From: mwestfal@csci.csusb.edu (Michael Westfall)
Date: Sat, 8 Jun 1996 19:22:48 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606090222.TAA20047@csci.csusb.edu>
Rendered body type: text/plain

> I'm just starting with DSP and thought (after creating some CW FIR
> filters) that I'd try to implement a 1200 baud packet TNC on the EZ-KIT
> Lite.
> I had assumed that implementing the modulation side of this was going to

> be easy but I've run into a major hurdle that I can't figure out.
>
> Generating the NRZI data and modulating it as a continuous phase FSK audio
> signal was easy. I have set up the DSP program so that it reads KISS
> packets from the RS-232 port and then generates the FSK. However, if I put
> the audio from the Kit into either my KAM or Tiny-2 neither of them will
> decode a packet.

I would guess what you are forgetting is to add the HDLC frame flags and
bit-stuffing....

>
> I'm lost. I know this is "back to basics" for most of you but has anyone
> any idea where I can start looking for the problem?

I'm no expert, so I'll defer to someone else to explain further...


```
--------------------------------------------------------------------------------
73 de Mike,      ax.25net:    N6KUY@W6JBT.#SOCA.CA.USA.NA
                  amprnet:    n6kuy@n6kuy.ampr.org [44.18.0.49]
                 internet :   mwestfal@csci.csusb.edu
                     web:     http://orion.csci.csusb.edu:8080
               "Linux: The Gates of Hell shall not prevail."
GCS/M { -d+ p+ c++ l u++ e+(*) m++(-) s/+ !n-(---) h-- !f g+ w+ t++ r-(--) y+ }
--------------------------------------------------------------------------------
```


39. Re: [HFSIG:1202] Re: Stuck at the beginning

From: Pete Hardie <hardie@duke.usask.ca>

Date: Sat, 8 Jun 1996 21:26:57 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.93.960608212140.22169A-100000@duke.usask.ca>
Rendered body type: text/plain

On Sat, 8 Jun 1996, Michael Westfall wrote:

> I would guess what you are forgetting is to add the HDLC frame flags and
> bit-stuffing....

That's all in there Mike because:
(a) I know they are in the DSP program.
(b) My "nrzi" program which decodes the signals not only decodes mine
correctly from the DSP but it also correctly decodes real packets off the
air. So my DSP has to be generating the flags and bit-stuffing otherwise
the "nrzi" program wouldn't decode it.

Thanks for the reply.

73 de Pete
ve5va.qrp@usask.ca

40. Re: [HFSIG:1201] Stuck at the beginning

From: Johan Forrer <forrerj@ucs.orst.edu>

Date: Sat, 08 Jun 1996 23:22:33 -0800
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19960609072233.38677264@ucs.orst.edu>
Rendered body type: text/plain

Hi Pete,

My two cent's worth:

>
>The fact that I can decode the NRZI output from the TNC modem eliminates
>several possible problems such as incorrect audio tones and the fact that
>the "nrzi" program can decode "real" packets using the modem's NRZI output
>suggests that the timing is also correct.
>

Seems like "nrzi" can decode both packets taken from your TNC's modem and
your DSP signal source - so that looks like the format is OK.

Have you looked at the audio levels going to the real TNC when you try your
DSP signal source? Perhaps they are overdriving the TNC. TNC's are quite
sensitive to the level and the fact that you see DCD comes on may not mean
too much. Perhaps some form of level adjustment may help. Otherwise, I would
defininately listen to that audio with an amplified speaker or scope and
determine whether is clean and without hum.

>I'm lost. I know this is "back to basics" for most of you but has anyone
>any idea where I can start looking for the problem?
>
>73 de Pete
>ve5va.qrp@usask.ca
>
>

Looks like you are well on your way.

--Johan

41. Re: [HFSIG:1204] Re: Stuck at the beginning

From: Pete Hardie <hardie@duke.usask.ca>
Date: Sun, 9 Jun 1996 19:23:56 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.93.960609191716.28968A-100000@duke.usask.ca>
Rendered body type: text/plain

On Sun, 9 Jun 1996, Johan Forrer wrote:

> Seems like "nrzi" can decode both packets taken from your TNC's modem and
> your DSP signal source - so that looks like the format is OK.
>
> Have you looked at the audio levels going to the real TNC when you try your
> DSP signal source? Perhaps they are overdriving the TNC.

I have checked that out and it doesn't make any difference. I wasn't sure
that it should because the signal that "nrzi" sees is what the modem has
decoded from the audio input. If the audio were overdriving the modem then
I would expect the program to see that as errors in the nrzi bitstream
coming out of the modem but "nrzi" has no trouble decoding the bitstream.

> Otherwise, I would
> defininately listen to that audio with an amplified speaker or scope and
> determine whether is clean and without hum.

I have listened to the audio and it sounds exactly like the packets I hear
on the air. I haven't got a scope but I'm going to try to borrow one to
have a look at the nrzi bitstream from the TNC's modem. Perhaps there are
subtle timing differences which don't show up due to the nature of my
sampling program.

Thanks Johan.

Pete

42. Re: [HFSIG:1200] Re: Further thoughts on pulse shaping for HF

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 10 Jun 1996 17:39:42 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606110039.RAA29160@servo.qualcomm.com>
Rendered body type: text/plain

While the usual methods for extracting timing were designed for
continuous streams of bits, I point out that just about all of our
applications now involve packetized data of one sort or another. So
that opens up some alternative methods for timing extraction, like a
one-shot scheme based on a sync vector in the front of the packet.

That is, instead of continuously recovering clock from the data
stream, you estimate it once during each packet and extrapolate it
from there. This is easy to do by sliding a correlator over the packet
-- when it hits its peak, you just count off the proper number of
samples per bit for the rest of the packet. Much easier.

With even cheap crystals being as good as they usually are, one-shot
timing schemes work pretty well as long as the packet size is limited
-- which it is anyway for many other reasons.

Too bad that one-shot schemes won't work for carrier recovery -- there's
too much doppler, noise and/or other impairments for this to work, so
you have to extract it as before with a Costas loop or something similar.

Phil

43. Re: [HFSIG:1206] Re: Further thoughts on pulse shaping for HF

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Wed, 12 Jun 1996 09:00:45 -0700 (PDT)
To: HFSIG@tapr.org
Message-ID: <Pine.OSF.3.91.960612084342.19054A-100000@ucs.orst.edu>
Rendered body type: text/plain

Hi Phil,

>
>
> While the usual methods for extracting timing were designed for
> continuous streams of bits, I point out that just about all of our
> applications now involve packetized data of one sort or another. So
> that opens up some alternative methods for timing extraction, like a
> one-shot scheme based on a sync vector in the front of the packet.

It is true that nearly all communications nowadays have some form of
header that allows for this kind of one-shot sync. Could you tell us a
bit more about the sync vector that you are now using. I did follow some
discussion a while ago about your explorations for a suitable sync vector.

I now have reasonable clock and data extraction working, but I am working
on extending the code to look for a sync vector. Although transmission of
the main data stream is DQPSK, I use ordinary DPSK for the sync vector.
I'll do as you suggest - obtain and set the clock at that point it
recognises the sync vector. My observations have been that there is a
fraction of a bit's worth slippage across my packets - not too much to be
of concern.

I also am considering using offset QPSK and was wondering; would the
extended Costas loop as for QPSK work for offset QPSK, or would I
need to change it?

>
>
>That is, instead of continuously recovering clock from the data
> stream, you estimate it once during each packet and extrapolate it
> from there. This is easy to do by sliding a correlator over the packet
> -- when it hits its peak, you just count off the proper number of
> samples per bit for the rest of the packet. Much easier.
>
> With even cheap crystals being as good as they usually are, one-shot
> timing schemes work pretty well as long as the packet size is limited
> -- which it is anyway for many other reasons.
>
> Too bad that one-shot schemes won't work for carrier recovery -- there's
> too much doppler, noise and/or other impairments for this to work, so
> you have to extract it as before with a Costas loop or something similar.
>
> Phil
>
>

The idea here is to use a number of orthogonal channels for data, one
channel will be unmodulated just to extract doppler.

The project has been a lot of fun but sure burns time and effort.
However, I'm learning a lot too.

--Johan

44. Re: [HFSIG:1207] Re: Further thoughts on pulse shaping for HF

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 12 Jun 1996 21:20:42 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606130420.VAA01103@servo.qualcomm.com>
Rendered body type: text/plain

>header that allows for this kind of one-shot sync. Could you tell us a
>bit more about the sync vector that you are now using. I did follow some

Well, I haven't settled on a final vector yet, but at the moment I'm
using a 26-bit vector constructed from the 13-bit Barker code (the
longest-known binary Barker code) with each bit Manchester encoded.
I.e., a "1" becomes "10" and a "0" becomes "01".

This does several things. It adds energy to the total vector, making

it easier to spot reliably, and it also adds two negative sidelobes
right next to the main positive lobe. In radar where you don't have
polarity information and are forced to look at signal envelopes, this
would be a big drawback. But this is actually an advantage to me since

I already have signal polarity from the carrier preamble, and the negative
sidelobes "sharpen" the main lobe, making it less likely for me to lock
one (or more) samples off the center of the main lobe.

I've simulated this vector pretty heavily and it seems adequate at the
nominal modem Eb/N0 of 3dB, but I wouldn't mind a little more margin.

Phil

45. Re: [HFSIG:1207] Re: Further thoughts on pulse shaping for HF

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 12 Jun 1996 21:30:22 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606130430.VAA01120@servo.qualcomm.com>
Rendered body type: text/plain

>I also am considering using offset QPSK and was wondering; would the
>extended Costas loop as for QPSK work for offset QPSK, or would I
>need to change it?

Yes, it works. The scheme I've been using is to sample both I&Q twice
per symbol, once at the center of the I symbol and again at the center
of the Q symbol. Two of these samples obviously become the recovered
data symbols. The out-of-phase samples (e.g., the one of the Q-channel
taken at the I-channel sampling instant) has its sign flipped by the
polarity of the current in-phase sample. Then the polarity corrected
out-of-phase Q-channel samples are subtracted from the polarity
corrected out-of-phase I-channel samples, producing the phase
residuals that my loop attempts to minimize in a least-squares
fashion.

This is basically the classic decision-directed 4-phase costas loop,
implemented in a sampled system.

I've been looking more closely at your exact question over the past
day or two. With my current scheme it *seems* that with SQPSK (but not
plain QPSK) there exist data patterns that produce no phase detector
output even when there is a carrier phase error. If you number the
four phase states 1-2-3-4, imagine an encoded phase sequence that goes
1-2-3-4-1-2-3-4 (or 4-3-2-1-4-3-2-1). (This corresponds roughly to an
unmodulated carrier offset above or below the real carrier). In this
case, all of the out-of-phase symbols are in the process of changing
when they're sampled by the in-phase symbol, leaving no residual even
when there's a carrier phase error.

The scheme still seems to work, at least for the random-like data
produced by the convolutional coder and interleaver, but this
phenomenon still bothers me. If nothing else, it means that the gain
of the "loop" (and the speed at which it converges) depends on the
signal pattern as well as the amplitude of the signal. Although this
doesn't actually affect the error performance of the modem (since I
operate in an a-posteriori batch mode where I repeatedly crunch each
block of symbols until I find the steady-state maximum likelihood
carrier phase and frequency), it does affect the cost in CPU cycles.

More when I figure this all out for myself.

Phil

46. Re: [HFSIG:1207] Re: Further thoughts on pulse shaping for HF

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 12 Jun 1996 21:34:19 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606130434.VAA01148@servo.qualcomm.com>
Rendered body type: text/plain

>The idea here is to use a number of orthogonal channels for data, one
>channel will be unmodulated just to extract doppler.

How well does the carrier reference channel work in practice, given
multipath? Can you really just lock a loop to it and derive carrier
phase for all your other channels, eliminating all the squaring (or
worse) losses inherent in suppressed carrier systems?

Phil

47. Re: [HFSIG:1210] Re: Further thoughts on pulse shaping for HF

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Thu, 13 Jun 1996 10:01:44 -0800
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19960613180144.3ff78574@ucs.orst.edu>
Rendered body type: text/plain

Phil,

At 11:49 PM 6/12/96 -0500, you wrote:
>>The idea here is to use a number of orthogonal channels for data, one
>>channel will be unmodulated just to extract doppler.
>
>How well does the carrier reference channel work in practice, given
>multipath? Can you really just lock a loop to it and derive carrier
>phase for all your other channels, eliminating all the squaring (or
>worse) losses inherent in suppressed carrier systems?
>
>Phil
>
>

That's a good question. Perhaps someone like Hakan, who are more familiar with MIL-STD 188 or STANAG modems could tell us more about that. I have seen the idea used for doppler compensation in both the 39 and 16 tone modems for this purpose. Perhaps to deal with a kind of doppler that affects the block of frequencies as a whole instead of only selective channels. As to how valid or effective that is, I really don't know.

Someone care to comment?

--Johan

48. Re: [HFSIG:1209] Re: Further thoughts on pulse shaping for HF

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Thu, 13 Jun 1996 10:01:38 -0800
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19960613180138.3ff7efd0@ucs.orst.edu>
Rendered body type: text/plain

Hi Phil,

At 11:49 PM 6/12/96 -0500, you wrote:
>>I also am considering using offset QPSK and was wondering; would the
>>extended Costas loop as for QPSK work for offset QPSK, or would I
>>need to change it?
>
>Yes, it works. The scheme I've been using is to sample both I&Q twice
>per symbol, once at the center of the I symbol and again at the center
>of the Q symbol. Two of these samples obviously become the recovered
>data symbols. The out-of-phase samples (e.g., the one of the Q-channel
>taken at the I-channel sampling instant) has its sign flipped by the
>polarity of the current in-phase sample. Then the polarity corrected
>out-of-phase Q-channel samples are subtracted from the polarity
>corrected out-of-phase I-channel samples, producing the phase
>residuals that my loop attempts to minimize in a least-squares
>fashion.

Thanks - good to know that.

>
>This is basically the classic decision-directed 4-phase costas loop,
>implemented in a sampled system.
>
>I've been looking more closely at your exact question over the past
>day or two. With my current scheme it *seems* that with SQPSK (but not
>plain QPSK) there exist data patterns that produce no phase detector
>output even when there is a carrier phase error. If you number the
>four phase states 1-2-3-4, imagine an encoded phase sequence that goes
>1-2-3-4-1-2-3-4 (or 4-3-2-1-4-3-2-1). (This corresponds roughly to an
>unmodulated carrier offset above or below the real carrier). In this
>case, all of the out-of-phase symbols are in the process of changing
>when they're sampled by the in-phase symbol, leaving no residual even
>when there's a carrier phase error.

I follow - interesting that SQPSK has that anomaly. I was going to ask whether you are using a scrambler as that would reduce the recurrence of

such a pattern, but your convolutional coding and interleaver will probably
have the same effect to randomize the pattern.

>
>The scheme still seems to work, at least for the random-like data
>produced by the convolutional coder and interleaver, but this
>phenomenon still bothers me. If nothing else, it means that the gain
>of the "loop" (and the speed at which it converges) depends on the
>signal pattern as well as the amplitude of the signal. Although this
>doesn't actually affect the error performance of the modem (since I
>operate in an a-posteriori batch mode where I repeatedly crunch each
>block of symbols until I find the steady-state maximum likelihood
>carrier phase and frequency), it does affect the cost in CPU cycles.

I played with a decision-directed scheme for a while that, at the middle of
the symbol, found the closest constellation point, and then determines
whether the received constellation needed to be rotated clockwise or
anti-clockwise. When the set of constellation points were {(1 0), (0 -1),
(-1 0), (0 -1)}, then either I or Q needed to be zero. One could then use
this non-zero offset as an (magnitude, direction) error estimate to be used
for loop control. This worked wonderfully until I started using RC pulse
shaping - the zero-crossing transitions for the RC pulses just played havoc.
At that point I switched to the extended Costas loop that took care of it,
however, as you said in an earlier posting, that type of loop drives I and Q
maximal and in my case, the transmitted and received constallations are
offset by a constant 45 degrees. That does not bother differential coding as
long as it is a stable lock, which is the case. I find that I can start the
demodulator midst of a random data sequence, obtain and hold lock - even for
33 second packets.

>
>More when I figure this all out for myself.
>
>Phil
>
>

Let us know the outcome - I'm curious.

--Johan

49. re pos of filter vs nyquist

From: zs6awk@global.co.za (Danie Brynard)
Date: Fri, 14 Jun 1996 23:49:42 +0200
To: hfsig@tapr.org
Message-ID: <199606150148.XAA27977@lin01.global.co.za>
Rendered body type: text/plain

I have been playing around with simulations to see if the position of a FIR
filter is important relative to the Nyquist frequency. Thus if I have a BPF
of say arbitrary passband and stopband does it matter where I place this
filter in the frequency range 0Hz to Fnyquist Hz ? (Fnyq=Fs/2)

According to my simulations it does not matter but I am not convinced. In
the analog world the rule is: the higher the freqency the more difficult it
is to make the same BPF.

Is wordlength or calculation precision perhaps the answer ? Is the number of
cycles per unit time of the wanted signal perhaps the answer ?

Could somebody give me a 'gut-feel' answer ? In other words a layman
explanation why it does/does not matter where one puts the filter.

I hope I explained myself well enough. Looking forward to all the
interesting replies :-)

73 danie zs6awk@global.co.za

50. Re: Further thoughts on pulse shaping for HF

From: "Bergzen Hakan, KARL" <hakan.bergzen@enator.se>
Date: Sat, 15 Jun 1996 16:39:00 +0100
To: hfsig <hfsig@tapr.org>
Message-ID: <31CDE3D2@noak>
Rendered body type: text/plain

Johan,

I have not been able to monitor this group for some time, just trying to

catch up, when I saw your reference to me. Even though I doubt I would have some information which not you and Phil already knows (I am very impressed of both of you being able to give the rest of us elaborate responses to our various questions, comments and ideas). This probably also goes for several silent readers of this group.

>>>The idea here is to use a number of orthogonal channels for data, one
>>>channel will be unmodulated just to extract doppler.
>>
>>How well does the carrier reference channel work in practice, given
>>multipath? Can you really just lock a loop to it and derive carrier
>>phase for all your other channels, eliminating all the squaring (or
>>worse) losses inherent in suppressed carrier systems?
>>
>That's a good question. Perhaps someone like Hakan, who are more familiar
>with MIL-STD 188 or STANAG modems could tell us more about that. I have
seen
>the idea used for doppler compensation in both the 39 and 16 tone modems
for
>this purpose. Perhaps to deal with a kind of doppler that affects the block
>of frequencies as a whole instead of only selective channels. As to how
>valid or effective that is, I really don't know.

The parallel tone modes within MIL-STD-188-110A use a special doppler correction tone, which will run continuously. It is the lowest tone in both modes (605 Hz for the 16-tone mode and 393.75 Hz for the 39-tone mode). It is only used for doppler correction, thus enabling the DSP to adjust the frequency offset for all data tones up or down correspondingly before symbol demodulation.

Both modes use BDPSK and QDPSK (depending on the data rate selected). At the lower data rates DPSK and in-band diversity combining are used. The 39-tone mode also use interleavers. Since only differential encoding is used the symbol decoding problem is more or less eliminated (I dont think it is possible to extract a carrier phase for one tone and use it for any other tone).

The protocol uses an initial preamble. The 39-tone mode use a 3-part preamble of different data tones all-in-all 23 signal elements (An extended preamble uses 97 signal elements). The data to be sent is divided into blocks. Each data block starts with a block sync. The block length depends on interleaver depth and data rate selected. A transmission is ended with an End-of-message sequence.

I think it would be a good idea to include a superb feature defined in the serial tone mode of the MIL-STD-188-110A, namely the Autobaud function. Within the initial preamble (and the re-occuring preamble enabling re-sync) the data rate and interleaver setting is encoded. This enables a receiving modem to automatically make the correct parameter setting and receive the message. The transmitting modem can by itself decide upon the most suitable data rate/interleaver depth.This makes it e.g. possible to easily construct adaptive ARQ schemes.

The trend in Europe and the U.S. seems to go from parallel tone modems to serial tone modems. They are more tricky to construct (including adaptive channel equalizers) but they show better performance because of their better utilization of the available channel (a sort of narrow-band SS, really). Though recent Australian studies (DSTO) using parallel tone modems together with trellis coding claims to perform better than serial tone modems.

For what it's worth...

Hakan   (hakan.bergzen@enator.se)

51. Re: [HFSIG:1190] Re: QPSK BANDWIDTH ?

From: Jim Sanford <JSANFORD@INFI.NET>
Date: Sat, 15 Jun 1996 21:04:37 -0400
To: hfsig@tapr.org
Message-ID: <199606160104.VAA28286@mh004.infi.net>
Rendered body type: text/plain

Phil
Forgive the long delay . .. the cost of my job which takes me
to sea . . .

Your comment about filters is part of why I'm hoping to do the thing
with simple chips and do ALL filtering in DSP. Of course, Ulrich Rhode
will remind me that a "ROOFING" filter would improve s/n in the IF . . .

Goal is acceptable performance at low cost and ease of duplication. I'm
hoping to turn this project into a kit for our club . . .

At 03:22 PM 6/4/96 -0500, you wrote:
>>   1. For what platform are you doing the OQPSK
>>satellite modem? I have 2 dsp-12's and would love to
>>beta test . . .
>
>The platform is a generic 486 or Pentium PC running my KA9Q NOS TCP/IP
>package. The modem will be integrated in as an interface driver for a
>SoundBlaster 16 card *without* any special DSP chips.
>
>Your work with the 'digital friendly' receiver sounds very
>interesting! I do hope that whatever you come up with can be easily
>replicated by the average amateur -- in my experience, that's where
>many promising amateur hardware projects fall down. It's the main
I absolutely agree...

>reason I've become so interested in using mass-market PC hardware to
>the greatest possible extent, doing everything else in software. But
>there are definite limits to the available hardware (particularly in
>the radios) so your work is most welcome. I agree, this stuff ought
>not to be as expensive and complicated as it seems to be.
I've been somewhat against using stuff in a PC because I hate the thought
of tying up a PC for a single function, but you've got a GREAT point
about the cost of mass market hardware vs anything for the much smaller
ham market. And, as costs of PC's drop . . . .

I'm already dedicating a 486/overdrive to the pacsats, so, what's another
cheap 486 on the network . . . <grin>

Thanks again for all you've done....73, Jim
wb4gcs

>The guy behind the Fox-Tango table, being used to selling very sharp
>and narrow crystal filters for HF contest and DX use, couldn't figure
>out why in the world I would want to make my rig go *wider*. Ah, the
>"narrowband mindset" seems pretty well entrenched. :-)
>

52. Re: [HFSIG:868] Re: DSP Tools

From: zs6awk@global.co.za (Danie Brynard)
Date: Sun, 16 Jun 1996 21:55:02 +0200
To: hfsig@tapr.org
Message-ID: <199606162353.VAA03929@lin01.global.co.za>
Rendered body type: text/plain

Phil is there perhaps a book that you can suggest if one wants to read more
about the inner working of the 486 and Pentuim and also on real-time
programming of it in assembler ? Something perhaps like 'asm programming on
the Pentuim for beginners' :-)

I am hooked onto the Motorola DSP56xxx family but is interested in your ideas..

danie zs6awk
zs6awk@global.co.za

>At 06:22 PM 2/8/96 -0600, you wrote:
>
>>As for DSP development platforms. Both TI (www.ti.com) and Motorola
>>(www.motorola.com) have inexpensive DSP Starter Kits (DSK) (TI) or
>>Evaluation Modules (EVM) (Motorola). The TI C5X DSK (p/n TMDS3200051) costs
>>$99.00 and the Motorola EVM560002 costs $149.00. I own the later and have
>>the former on order.
>
>Don't forget that you can do serious DSP on general purpose computers without
>a DSP board. Pentiums and the faster 486s rival many dedicated DSPs, especially
>on floating point computations.
>
>Phil
>
>

53. Re: [HFSIG:1216] Re: DSP Tools

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sun, 16 Jun 1996 14:51:19 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606162151.OAA15153@unix.ka9q.ampr.org>

Rendered body type: text/plain

Well, the standard Intel programmer's references are pretty good. They tell you all you really need to know to optimize DSP code on the Pentium, such as instruction clock counts and considerations on how to keep the pipelines full.

Phil

54. BTL performance tests

From: Robert.Glassey@nmp.nokia.com
Date: Mon, 17 Jun 96 10:17:35 +0100
To: hfsig@tapr.org
Message-ID: <H000029201ddaff7@MHS>
Rendered body type: text/plain

Hi all,

Over the weekend I did a few performance tests on my PC-Sound Blaster based DSP RTTY modem (BTL), and compared it with my PK232MBX. Here's the results. Any coments welcome.

	PK232MBX	BTL Ver 0.2
1 Noise performance: Eb/No for BER=10^-2	15dB	12dB
2 Adjacent channel rejection (+500Hz 98 baud RTTY)		
2a For 3dB rise in 10^-2 Eb/No:		
Square pulse interferer	9dB	30dB
Raised cosine pulse interferer	-	32dB
2b For clean wanted signal, 10^-2 BER:		
Square pulse interferer	16dB	32dB
Raised cosine pulse interferer	-	35dB

Observations:

1. The BER taken includes errors due to both sync errors and data bit errors. For both modems relative contributions were about 50-50. If a bit synchronous mode were used rather than asynchronous baudot, BER would be half. My measurements show this is equivelent to a 1dB improvement in Eb/No for both modems.

2. Two different interferer types were used. Both were phase continous FSK, 98 baud baudot, centred 500Hz above the wanted signal (wanted 1275/1445, interferer 1775/1945) Standard RTTY has a square pulse shape resulting in high adjacent channel power. (-30dB @ 500Hz) This limited measurements (and is a real on air limit too) However I tried shaping the interferering FSK pulses with raised cosine edges (shaped 64 of 82 samples). ACP was reduced to -45dB @ 500Hz. Using this interferer, the true performance of the modem could be determined. This shaping reduced the interference RMS level to 72% and this was taken into account.

   The minimum signal level without intefernce was found to be 1 ADC step peak to peak, with dithering from the noise of almost 3 ADC steps peak-peak. (signals were calculated in 16 bits then quantised to 8 bits to allow noise dithering). At this level performance was still 12dB Eb/No @ BER 10^-2. This minimum signal level was 44dB below the inteference level used above. Thus quantisation was not a limiting factor in the above measurements. Lower levels were possible but Eb/No was worsened and quantisation boundries became significant.

Measurement Technique:

1. Eb/No measurements

Sampling rate was 8KHz

The signal was 45 baud Baudot, phase continuous AFSK, 1275Hz

space and 1445Hz mark (170Hz shift). RMS to peak ratio =
0.71

The noise was generated by a 24bit maximum length PRN
generator (poly=0x8D0000), doing 8 shifts per sample to get
an 8 bit uniform random noise source. RMS to peak ratio of a
uniform random noise source = 0.58

I have assumed a uniform distribution is OK since after
bandpass filtering and integration the noise distribution
aproaches gausian. (roughly speaking, the noise is averaged
over the bit period, ie 178 samples, making the noise
gausian by summation)

For Eb/No calculations I have used the RMS levels at the
8kHz sample rate.

Peak-peak levels of signal=45 and noise=128 becomes 12db
Eb/No thus:

```
Eb/No =  20*log10 ((45/2 * 0.71) / (128/2 * 0.58))
         + 10*log10(4000 Hz) - 10*log10(45 BPS)
      =  12.2 dB
```

2. Adjacent channel measurements

   The signals are described above.

   The interferer baud rate of 98 baud was chosen to try to
   ovoid synchronisation between the wanted and interferer
   so errors would appear more random, hopefully giving a
   more accurate BER. However it does make the interferer
   wider, but pulse shaping dealt with that. This may not
   be the best method.

   Results are given for two different cases. 1. For a
   3dB rise in the Eb/No required for 10^-2 BER, and
   2. The interference level that gives a BER of 10^-2
   when the wanted signal is clean. These figures show
   respectivly the degradation in weak signal performance,
   and the maximum tolerable adjacent channel signal level
   when the wanted signal is clean.

So, 12dB Eb/No doesn't sound all that hot, does it? But is this what
should be expected for FSK without ECC? Eb/No would be 11dB if a bit
synchronous mode was used. These are for a BER of 10^-2.

Cheers,

Rob

55. Re: [HFSIG:1209] Re: Further thoughts on pulse shaping for HF

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Mon, 17 Jun 1996 03:16:05 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606171016.DAA00188@unix.ka9q.ampr.org>
Rendered body type: text/plain

FYI, more about offset QPSK vs conventional QPSK.

It looks as though the advantages of staggered (offset) QPSK are
outweighted by the disadvantages, so I've switched to straight QPSK.

There are two main advantages of offset QPSK: 1) less envelope droop
when the signal is bandlimited, which turns into less sideband
regrowth when the signal is nonlinearly amplified, and 2) less
crosstalk between channels when the carrier phase reference is
inaccurate.

I originally chose SQPSK for reason #2; I don't really care too much
about #1 since I am going to operate in a power-controlled regime
where I assume there'll be a fair amount of transmit headroom most of
the time, hence plenty of linearity. And even if there were sideband
regrowth it wouldn't be as important a source of interference since
I'm already operating at very low S/N ratios thanks to power control
and coding.

But it turns out that the crosstalk resistance of SQPSK is largely mitigated by greatly increased pattern-sensitive jitter in the carrier recovery system. In other words, although SQPSK has increased carrier phase jitter tolerance, it inherently increases the jitter of the recovered carrier which squanders the improved performance.

I couldn't find a precise quantitative formulation for this, but from what I saw it looks like any gains SQPSK might have (especially at the very low Es/N0s I'm using) are small or even negative. And the last straw was that SQPSK is harder to implement than straight QPSK. So I've switched to straight QPSK.

Phil

56. Re: [HFSIG:1212] Re: Further thoughts on pulse shaping for HF

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Mon, 17 Jun 1996 03:47:44 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606171047.DAA00230@unix.ka9q.ampr.org>
Rendered body type: text/plain

>I follow - interesting that SQPSK has that anomaly. I was going to ask
>whether you are using a scrambler as that would reduce the recurrence of
>such a pattern, but your convolutional coding and interleaver will probably
>have the same effect to randomize the pattern.

Re the anomaly I described (certain data patterns killing the SQPSK phase detector) I'm now not exactly sure that this is real. But it *does* seem from reading the books that there is a pattern-sensitive phase jitter in SQPSK that's not present in plain QPSK, and the problem gets worse when you tightly filter the signal. So that's why I've switched from SQPSK to plain QPSK. Simplified the code, too.

Re scramblers, no, I'm not scrambling. Yes, the convolutional coding and interleaving does a pretty good job. It certainly sounds noiselike except when I send a big block of zeroes, which come out as a tone. But the loop can track this fine anyway (remember I'm doing one-shot symbol timing up front).

Re your discussion of decision-directed carrier recovery, it works great when the Es/N0 is high enough to work without coding. But at the Es/N0 ratios I'm using with my coding, there are plenty of errors in the decisions because they're made by simple slicing, without benefit of any error correction. So many of the phase detector outputs are wrong (though not most, since the error rate is still less than 50%). This adds noise to the feedback loop over and above the noise in the input signal. In the literature this is called "squaring loss" (although strictly speaking that applies only to BPSK -- in QPSK it should be called "4th power loss"). Squaring loss is a function of loop design, Es/N0 and of the loop signal-to-noise ratio.

Squaring loss forces you to use a much narrower loop filter than you'd otherwise use in order to maintain an acceptably high S/N ratio within the loop. That means you get pretty sensitive to doppler.

Another good illustration of how trying to conserve bandwidth (e.g., by using QPSK instead of BPSK in the first place) often ends up forcing you to use more power per bit even when it shouldn't -- ideal QPSK and BPSK (with perfect carrier recovery) have exactly the same Eb/N0 requirements. I'm still having a hard time getting to within 1 dB of the theoretical code performance in my QPSK modem.

Phil

57. Sinewave generator source code

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Mon, 17 Jun 1996 08:16:07 -0400
To: hfsig@tapr.org, ss@tapr.org
Message-ID: <1C54EE50.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

For what its worth, I found the article I was looking for (Radio Electronics, October 1991). The author used a C program to generate the data for the EPROM feeding into the DAC. His source code is shown below. I also attached a sample output file to this note. The EPROM is a standard 2716. Of course, the code can be modified to suit your needs.

I intend to use this to produce a LUT (Look Up Table) inside a Xilinx FPGA. I want to use the LUT as the basis for a QPSK modulator. If I can feed serial data into the Xilinx, I can then produce I and Q components and use these as inputs into a custom up/down counter. The output of the counter will feed into the LUT. The output of the LUT would then feed into an external DAC. I'm not sure if this can work, but I will anyone who cares about this now what results I came up with.

73s de
Tony, KE4ATO

```
/* This program calculates the value of the sine function
   offset so that the 4th and 1st quadrants cause a code
   from 0 to 255. Code is generated to fill a 2048 byte prom
   (2716 or equivalent) for a full circle of 2*pi radians.
   Other size memory may be used by changing the value of
   bytes in the declaration table.
*/

#include <stdio.h>
#include <math.h>

main ()
{
  double p=0;           /* phase input to sine function */
       double S=0;      /* output value of the true sine function */
       int s;           /* amplitude truncated to 8 bits */
       double sin();    /* true sine function */
       double pi=3.141592654;
       int addr=0;      /* address of EPROM */
       int bytes=2048;  /* size of EPROM in bytes */

       printf("          0    1    2    3    4    5    6    7    8");
       printf("    9    A    B    C    D    E    F  \n");
       while (addr < bytes)
         {
         if (addr % 16 == 0)
           printf("\n%4x   ", addr);
         p = 2.0*pi*( (double) addr)/( (double) bytes);
         S = 127.5*(1.0 + sin(p - pi/2.0));       /* gives 0 at -90 deg */
         s = ( (int) S);                          /* convert to integer */
         if (S - ( (double) s) >= 0.5)            /* rounds if necessary */
           s++;
         printf("   %2x", s);
         addr++;                                  /* increment address */
         }
     return (0);
     }
```

[Attachment: output.txt; text/basic; 9,290 bytes]

58. Re: [HFSIG:1218] BTL performance tests

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Mon, 17 Jun 1996 08:40:14 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960617083834.20926A-100000@ucs.orst.edu>
Rendered body type: text/plain

Hi Robert,

Congratulations! - Nice writeup.

Is the program generally available yet, or what are your future plans.

Keep up the good work.

--Johan

On Mon, 17 Jun 1996 Robert.Glassey@nmp.nokia.com wrote:

> Hi all,
>
> Over the weekend I did a few performance tests on my PC-
> Sound Blaster based DSP RTTY modem (BTL), and compared it
> with my PK232MBX. Here's the results. Any coments welcome.

>
>   PK232MBX   BTL Ver 0.2
>
> 1 Noise performance: 15dB   12dB
>   Eb/No for BER=10^-2
>
> 2 Adjacent channel rejection (+500Hz 98 baud RTTY)
>
> 2a For 3dB rise in 10^-2 Eb/No:
>   Square pulse interferer   9dB   30dB
>   Raised cosine pulse interferer   -   32dB
>
> 2b For clean wanted signal, 10^-2 BER:
>   Square pulse interferer   16dB   32dB
>   Raised cosine pulse interferer -   35dB
>
>
> Observations:
>
> 1. The BER taken includes errors due to both sync
>   errors and data bit errors. For both modems
>   relative contributions were about 50-50.
>   If a bit synchronous mode were used rather than
>   asynchronous baudot, BER would be half. My
>   measurements show this is equivelent to a 1dB
>   improvement in Eb/No for both modems.
>
> 2. Two different interferer types were used. Both were
>   phase continous FSK, 98 baud baudot, centred 500Hz
>   above the wanted signal (wanted 1275/1445, interferer
>   1775/1945) Standard RTTY has a square pulse shape
>   resulting in high adjacent channel power. (-30dB @
>   500Hz) This limited measurements (and is a real on air
>   limit too) However I tried shaping the interferering
>   FSK pulses with raised cosine edges (shaped 64 of 82
>   samples). ACP was reduced to -45dB @ 500Hz. Using this
>   interferer, the true performance of the modem could be
>   determined. This shaping reduced the interference RMS
>   level to 72% and this was taken into account.
>
>   The minimum signal level without intefernce was found
>   to be 1 ADC step peak to peak, with dithering from the
>   noise of almost 3 ADC steps peak-peak. (signals were
>   calculated in 16 bits then quantised to 8 bits to allow
>   noise dithering). At this level performance was still
>   12dB Eb/No @ BER 10^-2. This minimum signal level was
>   44dB below the inteference level used above. Thus
>   quantisation was not a limiting factor in the above
>   measurements. Lower levels were possible but Eb/No was
>   worsened and quantisation boundries became significant.
>
>
> Measurement Technique:
>
> 1. Eb/No measurements
>
> Sampling rate was 8KHz
>
> The signal was 45 baud Baudot, phase continuous AFSK, 1275Hz
> space and 1445Hz mark (170Hz shift). RMS to peak ratio =
> 0.71
>
> The noise was generated by a 24bit maximum length PRN
> generator (poly=0x8D0000), doing 8 shifts per sample to get
> an 8 bit uniform random noise source. RMS to peak ratio of a
> uniform random noise source = 0.58
>
> I have assumed a uniform distribution is OK since after
> bandpass filtering and integration the noise distribution
> aproaches gausian. (roughly speaking, the noise is averaged
> over the bit period, ie 178 samples, making the noise
> gausian by summation)
>
> For Eb/No calculations I have used the RMS levels at the
> 8kHz sample rate.
>
> Peak-peak levels of signal=45 and noise=128 becomes 12db
> Eb/No thus:
>

> Eb/No = 20*log10 ((45/2 * 0.71) / (128/2 * 0.58))
>   + 10*log10(4000 Hz) - 10*log10(45 BPS)
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>
>
> 2. Adjacent channel measurements
>
>   The signals are described above.
>
>   The interferer baud rate of 98 baud was chosen to try to
>   ovoid synchronisation between the wanted and interferer
>   so errors would appear more random, hopefully giving a
>   more accurate BER. However it does make the interferer
>   wider, but pulse shaping dealt with that. This may not
>   be the best method.
>
>   Results are given for two different cases. 1. For a
>   3dB rise in the Eb/No required for 10^-2 BER, and
>   2. The interference level that gives a BER of 10^-2
>   when the wanted signal is clean. These figures show
>   respectivly the degradation in weak signal performance,
>   and the maximum tolerable adjacent channel signal level
>   when the wanted signal is clean.
>
>
>
> So, 12dB Eb/No doesn't sound all that hot, does it? But is this what
> should be expected for FSK without ECC? Eb/No would be 11dB if a bit
> synchronous mode was used. These are for a BER of 10^-2.
>
> Cheers,
>
> Rob
>
>
>
>
>
>

59. Re: [HFSIG:1219] Re: Further thoughts on pulse shaping for HF

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Mon, 17 Jun 1996 09:09:34 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.91.960617084032.20926B-100000@ucs.orst.edu>
Rendered body type: text/plain

Hi Phil,

On Mon, 17 Jun 1996, Phil Karn wrote:

> FYI, more about offset QPSK vs conventional QPSK.
>
> It looks as though the advantages of staggered (offset) QPSK are
> outweighted by the disadvantages, so I've switched to straight QPSK.
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> There are two main advantages of offset QPSK: 1) less envelope droop
> when the signal is bandlimited, which turns into less sideband
> regrowth when the signal is nonlinearly amplified, and 2) less
> crosstalk between channels when the carrier phase reference is
> inaccurate.
>
> I originally chose SQPSK for reason #2; I don't really care too much
> about #1 since I am going to operate in a power-controlled regime
> where I assume there'll be a fair amount of transmit headroom most of
> the time, hence plenty of linearity. And even if there were sideband
> regrowth it wouldn't be as important a source of interference since
> I'm already operating at very low S/N ratios thanks to power control
> and coding.
>
> But it turns out that the crosstalk resistance of SQPSK is largely
> mitigated by greatly increased pattern-sensitive jitter in the carrier
> recovery system. In other words, although SQPSK has increased carrier
> phase jitter tolerance, it inherently increases the jitter of the
> recovered carrier which squanders the improved performance.
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> I couldn't find a precise quantitative formulation for this, but from
> what I saw it looks like any gains SQPSK might have (especially at the

> very low Es/N0s I'm using) are small or even negative. And the last
> straw was that SQPSK is harder to implement than straight QPSK. So
> I've switched to straight QPSK.
>
> Phil
>
>
>
>

Your initial considerations for using the offset format seems quite valid
if one perhaps can overcome the side effects you mention.

About QPSK: The clean compact signal format that I see here is very
encouraging. Using raised cosine pulse shaping is a bit of a hassle, but
well worth it. I think if one can address the issue of non-linear
amplification, that this may not be a bad way to go.

One other thing that I experimented with this weekend was
non-coherent detection of DQPSK. After getting it to work, I was quite
surprized at how well it worked out. The constellations were tight
and text-book style. There is the expected recovery loss (i.e. lower
dynamic range) by giving up carrier extraction, however, I am of the
opinion that in the end it may work out best for HF especially
when used in conjunction with good coding.

The issue about the use of an unmodulated carrier for doppler compensation:
One of the first things that I do in my modem is to create an analytic
signal, with a frequency shifter stage built into the front end. The
frequency shifter is controlled by a doppler extractor algorithm.
Doppler estimation is modelled as a second order AR process derived from
the special unmodulated carrier after filtering by a narrow-band filter.
The frequency estimate is obtained solving two simulataneous equations
(Yule-Walker). There are provisions to deal with times when the carrier has
faded - pretty much a very heavy damped system similar to what I do with
clock extraction.

Good luck with your project.

--Johan

60. Re: [HFSIG:1218] BTL performance tests

From: N0AOT@aol.com
Date: Mon, 17 Jun 1996 14:02:57 -0400
To: hfsig@tapr.org, Robert.Glassey@nmp.nokia.com
Message-ID: <960617140256_558017278@emout09.mail.aol.com>
Rendered body type: text/plain

Hi Bob--

In a message dated 96-06-17 06:48:58 EDT, you write:

>Hi all,
>
>Over the weekend I did a few performance tests on my PC-
>Sound Blaster based DSP RTTY modem (BTL), and compared it
>with my PK232MBX. Here's the results. Any coments welcome.
>
>

Very interesting weekend project you had there! I am not familiar with the
BTL software you mentioned. Who is the author, and where is it available? Can
the hardware interupt and io address of the sound card be changed so that it
can be run on a notebook computer that has a "Sound Blaster compatible"
chip?

I have a Motorola DSP56002EVM board, and would like to run similar tests...

--73's de Bob Carlson, n0aot@amsat.org

61. Re: [HFSIG:1222] Re: BTL performance tests

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Mon, 17 Jun 1996 14:56:48 -0400
To: hfsig@tapr.org
Message-ID: <1C5AC3F0.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

Johan,

I think you have me confused with someone else. I didn't write the
text below - wish I did though. Its pretty impressive!

73s de
Tony, KE4ATO

P.S. My first name is Robert, but I go by Tony (middle name Anthony)
in all of my correspondence.

_____ Reply Separator _____
Subject: [HFSIG:1222] Re: BTL performance tests
Author: hfsig@tapr.org at BALT.SMTP
Date: 6/17/96 10:49 AM

Hi Robert,

Congratulations! - Nice writeup.

Is the program generally available yet, or what are your future plans.

Keep up the good work.

--Johan

On Mon, 17 Jun 1996 Robert.Glassey@nmp.nokia.com wrote:

> Hi all,
>
> Over the weekend I did a few performance tests on my PC-
> Sound Blaster based DSP RTTY modem (BTL), and compared it
> with my PK232MBX. Here's the results. Any coments welcome.
>
>   PK232MBX   BTL Ver 0.2
>
> 1 Noise performance: 15dB   12dB
>   Eb/No for BER=10^-2
>
> 2 Adjacent channel rejection (+500Hz 98 baud RTTY)
>
> 2a For 3dB rise in 10^-2 Eb/No:
>   Square pulse interferer   9dB   30dB
>   Raised cosine pulse interferer   -   32dB
>
> 2b For clean wanted signal, 10^-2 BER:
>   Square pulse interferer   16dB   32dB
>   Raised cosine pulse interferer -   35dB
>
>
> Observations:
>
> 1. The BER taken includes errors due to both sync
>   errors and data bit errors. For both modems
>   relative contributions were about 50-50.
>   If a bit synchronous mode were used rather than
>   asynchronous baudot, BER would be half. My
>   measurements show this is equivelent to a 1dB
>   improvement in Eb/No for both modems.
>
> 2. Two different interferer types were used. Both were
>   phase continous FSK, 98 baud baudot, centred 500Hz
>   above the wanted signal (wanted 1275/1445, interferer
>   1775/1945) Standard RTTY has a square pulse shape
>   resulting in high adjacent channel power. (-30dB @
>   500Hz) This limited measurements (and is a real on air
>   limit too) However I tried shaping the interferering
>   FSK pulses with raised cosine edges (shaped 64 of 82
>   samples). ACP was reduced to -45dB @ 500Hz. Using this
>   interferer, the true performance of the modem could be
>   determined. This shaping reduced the interference RMS
>   level to 72% and this was taken into account.
>
>   The minimum signal level without intefernce was found
>   to be 1 ADC step peak to peak, with dithering from the
>   noise of almost 3 ADC steps peak-peak. (signals were
>   calculated in 16 bits then quantised to 8 bits to allow

>   noise dithering). At this level performance was still
>   12dB Eb/No @ BER 10^-2. This minimum signal level was
>   44dB below the inteference level used above. Thus
>   quantisation was not a limiting factor in the above
>   measurements. Lower levels were possible but Eb/No was
>   worsened and quantisation boundries became significant.
>
>
> Measurement Technique:
>
> 1. Eb/No measurements
>
> Sampling rate was 8KHz
>
> The signal was 45 baud Baudot, phase continuous AFSK, 1275Hz
> space and 1445Hz mark (170Hz shift). RMS to peak ratio =
> 0.71
>
> The noise was generated by a 24bit maximum length PRN
> generator (poly=0x8D0000), doing 8 shifts per sample to get
> an 8 bit uniform random noise source. RMS to peak ratio of a
> uniform random noise source = 0.58
>
> I have assumed a uniform distribution is OK since after
> bandpass filtering and integration the noise distribution
> aproaches gausian. (roughly speaking, the noise is averaged
> over the bit period, ie 178 samples, making the noise
> gausian by summation)
>
> For Eb/No calculations I have used the RMS levels at the
> 8kHz sample rate.
>
> Peak-peak levels of signal=45 and noise=128 becomes 12db
> Eb/No thus:
>
> Eb/No = 20*log10 ((45/2 * 0.71) / (128/2 * 0.58))
>   + 10*log10(4000 Hz) - 10*log10(45 BPS)
>   = 12.2 dB
>
>
> 2. Adjacent channel measurements
>
>   The signals are described above.
>
>   The interferer baud rate of 98 baud was chosen to try to
>   ovoid synchronisation between the wanted and interferer
>   so errors would appear more random, hopefully giving a
>   more accurate BER. However it does make the interferer
>   wider, but pulse shaping dealt with that. This may not
>   be the best method.
>
>   Results are given for two different cases. 1. For a
>   3dB rise in the Eb/No required for 10^-2 BER, and
>   2. The interference level that gives a BER of 10^-2
>   when the wanted signal is clean. These figures show
>   respectivly the degradation in weak signal performance,
>   and the maximum tolerable adjacent channel signal level
>   when the wanted signal is clean.
>
>
>
> So, 12dB Eb/No doesn't sound all that hot, does it? But is this what
> should be expected for FSK without ECC? Eb/No would be 11dB if a bit
> synchronous mode was used. These are for a BER of 10^-2.
>
> Cheers,
>
> Rob
>
>
>
>
>
>

62. Re: [HFSIG:1224] Re: BTL performance tests

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Mon, 17 Jun 1996 17:19:52 -0400

To: hfsig@tapr.org
Message-ID: <1C5CC660.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

I am not the author of this software project! I don't know how my name
got mixed up.

Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1224] Re: BTL performance tests
Author: hfsig@tapr.org at BALT.SMTP
Date: 6/17/96 1:15 PM

Hi Bob--

In a message dated 96-06-17 06:48:58 EDT, you write:

>Hi all,
>
>Over the weekend I did a few performance tests on my PC-
>Sound Blaster based DSP RTTY modem (BTL), and compared it
>with my PK232MBX. Here's the results. Any coments welcome.
>
>

Very interesting weekend project you had there! I am not familiar with the
BTL software you mentioned. Who is the author, and where is it available? Can
the hardware interupt and io address of the sound card be changed so that it
can be run on a notebook computer that has a "Sound Blaster compatible"
chip?

I have a Motorola DSP56002EVM board, and would like to run similar tests...

--73's de Bob Carlson, n0aot@amsat.org

63. (no subject)

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 17 Jun 1996 20:37:10 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606180337.UAA02588@servo.qualcomm.com>
Rendered body type: text/plain

Subject: Re: [HFSIG:1223] Re: Further thoughts on pulse shaping for HF

Johan,

Noncoherent schemes have a lot going for them on fading, dispersive
channels. Not only because it's hard to extract a carrier phase
that's rapidly changing, but also because the dynamic range of the
fading tends to swamp the relatively small Eb/N0 ratio differences
between coherent and noncoherent modulation that are otherwise
important on severely power-constrained nonfading AWGN channels.

The three most important things on a fading channel are diversity,
diversity and diversity. :-) Diversity can be in time, space or
frequency. Coding helps promotes diversity, particularly in time by
spreading out the influence of one data bit over many symbols (only
some of which may get through). And even QPSK can help by simply
creating more slots for those coded symbols in the first place.

Phil

64. Re: [HFSIG:1223] Re: Further thoughts on pulse shaping for HF

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 17 Jun 1996 20:57:35 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606180357.UAA02630@servo.qualcomm.com>
Rendered body type: text/plain

>One of the first things that I do in my modem is to create an analytic
>signal, with a frequency shifter stage built into the front end. The
>frequency shifter is controlled by a doppler extractor algorithm.

This is exactly my approach also. The frequency shifter is at present
set to a fixed value based on the estimated frequency of the carrier
burst in the packet preamble. This assumes that any residual error is

small compared to the data rate (it is), so it can be tracked out
after filtering and downsampling from 8x to 1x the symbol rate. (The
downsampling starts once I have obtained one-shot symbol sync from the
preamble).

Although all further processing has to be done with complex
arithmetic, performance-wise I think I'm still ahead after the
downsampling.

Eventually I will probably have to add a doppler chirper to the
frequency shifter driven by an orbit tracking routine. This is
relatively straightforward given accurate orbital elements, time and
station location.

Phil

65. Re: [HFSIG:1225] Re: BTL performance tests

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Mon, 17 Jun 1996 21:54:58 -0800
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19960618055458.4c9f67fc@ucs.orst.edu>
Rendered body type: text/plain

At 02:21 PM 6/17/96 -0500, you wrote:
>   Johan,
>
>   I think you have me confused with someone else. I didn't write the
>   text below - wish I did though. Its pretty impressive!
>
>   73s de
>   Tony, KE4ATO
>
>   P.S. My first name is Robert, but I go by Tony (middle name Anthony)
>   in all of my correspondence.
>
>

Strange things do happen :-)

My reply was directed at Robert (Rob) Glassey's fine work (please see below)
- somehow we got our lines crossed - no harm done. Still a nice piece of
work Rob.

--Johan

>Hi Robert,
>
>Congratulations! - Nice writeup.
>
>Is the program generally available yet, or what are your future plans.
>
>Keep up the good work.
>

...... some lines deleted

>
>
>On Mon, 17 Jun 1996 Robert.Glassey@nmp.nokia.com wrote:
>

...... some more lines deleted

66. [HFSIG:1222] Re: BTL performance tests

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 18 Jun 96 10:15:50 +0100
To: hfsig@tapr.org
Message-ID: <H000029201dfc18f@MHS>
Rendered body type: text/plain

> Hi Robert,
>
> Congratulations! - Nice writeup.
>
> Is the program generally available yet, or what are your future plans.
>
> Keep up the good work.
>

> --Johan
>
>
> On Mon, 17 Jun 1996 Robert.Glassey@nmp.nokia.com wrote:
>
> > Hi all,
> >
> > Over the weekend I did a few performance tests on my PC-
> > Sound Blaster based DSP RTTY modem (BTL), and compared it
> > with my PK232MBX. Here's the results. Any coments welcome.

Thanks, Its still on beta test, with V0.2beta out very soon (I hope)
- Sorry to those testers still waiting for the next version,
unfortunatly my weekend projects don't always get the time they deserve.

The release version 1.0, should be ready in a copule of weeks. This
version recieves standard baudot only, with variable baud, shift and
centre frequency, plus a few other features.

My next release (could be 6 months away) will have TX, and a wide shift
mode for comercial news services etc. I hope to include AMTOR FEC and
ARQ modes as well. Plus any improvements in performance I can find,
probably a bit-synchronous demod mode (for standard 7.5 bit baudot), and
maybe an improved tuning indicator, auto tuning, and mode/baud
detection.

This was my intention from the start for a first release, but things
have taken a while, and I think people want it as it is right now.

Ultimatly I would like to use BTL as a platform for more complex modes
such as PACTOR II or Clover II or other multicarrier modes that may be
developed through HFSIG. I'm also keen to implement very low SNR modes,
such as DBPSK with heaps of ECC using a type II hybrid ARQ protocol.

I intend to release up to version 2 as public domain freeware.

Cheers,

Rob Glassey, G0VTQ

67. Well, the standard Intel programmer's references are pretty good. They tell you all you really need

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Tue, 18 Jun 1996 11:26:45 +0200
To: hfsig@tapr.org
Message-ID: <s1c693f3.043@KIDD.CO.ZA>
Rendered body type: text/plain

Well, the standard Intel programmer's references are pretty good. They tell you all you really need to know to
optimize DSP code on the
Pentium, such as instruction clock counts and considerations on how to keep the pipelines full.

Phil

I see. Thanks. 73 danie

68. [HFSIG:1224] Re: BTL performance tests

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 18 Jun 96 10:45:45 +0100
To: hfsig@tapr.org
Message-ID: <H000029201dfdd45@MHS>
Rendered body type: text/plain

> Hi Bob--
>
> In a message dated 96-06-17 06:48:58 EDT, you write:
>
> >Hi all,
> >
> >Over the weekend I did a few performance tests on my PC-
> >Sound Blaster based DSP RTTY modem (BTL), and compared it
> >with my PK232MBX. Here's the results. Any coments welcome.
> >
> >
>
> Very interesting weekend project you had there! I am not familiar

> with the BTL software you mentioned. Who is the author, and where is
> it available? Can the hardware interupt and io address of the sound
> card be changed so that it can be run on a notebook computer that has
> a "Sound Blaster compatible" chip?
>
> I have a Motorola DSP56002EVM board, and would like to run similar
> tests...
>
> --73's de Bob Carlson, n0aot@amsat.org

Hi Bob,

BTL is not on general release yet, but soon will be. BTL is my 'baby'
and has taken me since around August last year to write (very slowly
since its a weekend project). When released, I hope to get it onto an
FTP site somewhere, maybe the TAPR site? This version is FREE!

Soundblaster parameters are user configurable, and it accepts the
standard sound blaster enviroment variable BLASTER=....

It should work on any soundblaster compatible from the original v1.0 to
the latest AWE32.

It should run on your notebook if its fast enough. To date, the slowest
machine I've tried is a 486sx25 and that works fine. If speed is a
problem I may release a cut down 1.5 version which should run faster.

The tests were done with a C program that generated test signals plus
noise that I used as my signal generator for the tests.

The test signal was played on the SB for the PK 232, and taken directly
from disk, in real time, for the BTL tests, by substituting the
soundblaster DMA for a double buffered disk routine.

I also have a DAT recorder that I have used in the past for testing.
I'll have to repeat these BTL tests using recorded signals.

Cheers,

Rob, G0VTQ

69. (no subject)

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Wed, 19 Jun 1996 15:46:16 +0200
To: hfsig@tapr.org
Message-ID: <s1c82242.070@KIDD.CO.ZA>
Rendered body type: text/plain

I also have a DAT recorder that I have used in the past for testing.
I'll have to repeat these BTL tests using recorded signals.

Bob how well does the DAT recorder work and is it not too expensive ? I also have a need for soemthing
like that.
It is difficult with the sat signals as they pass very quickly...no time to recompile and download hi
:-)

danie zs6awk

70. [HFSIG:1233]

From: Robert.Glassey@nmp.nokia.com
Date: Wed, 19 Jun 96 16:30:26 +0100
To: hfsig@tapr.org
Message-ID: <H000029201e2ac42@MHS>
Rendered body type: text/plain

>> I also have a DAT recorder that I have used in the past for testing.
>> Bob how well does the DAT recorder work and is it not too expensive ?
> I also have a need for soemthing like that.
> It is difficult with the sat signals as they pass very quickly...no
> time to recompile and download hi :-)
>
> danie zs6awk

Hi Danie,

They work very well, even preserve timing on amtor/pactor. They are a
little pricy, mine cost around $650 US, but it is a walkman model, (Sony
TCD-D7). A hi-fi component style version should be cheaper. I find the

walkman handy for taking a known test signal from one PC to another for testing on different sound cards etc. They are also usefull for recording a live QSO then replaying it over and over while modifying the program to test with a known signal. They are also available second hand occasionally, but beware of older models, reliability was a problem with them, and they're probably on their last legs now if they haven't already died.

If you have a sound blaster your best bet (and cheapest) would be to record the satilite on disk the play it back to the DSP. 10 minutes will take about 5MB. You may need make your DSP code more independent of the PC (buffer output etc) or get another PC!

Its actually quite simple to write a C program to play the sound file and act as a dumb terminal at the same time.

73's

Rob,

G0VTQ

71. Re: [HFSIG:1228] Re: Further thoughts on pulse shaping for HF

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Wed, 19 Jun 1996 09:18:12 -0800
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19960619171812.3f8ffe56@ucs.orst.edu>
Rendered body type: text/plain

Phil,

.....

>
>Although all further processing has to be done with complex
>arithmetic, performance-wise I think I'm still ahead after the
>downsampling.
>
>Eventually I will probably have to add a doppler chirper to the
>frequency shifter driven by an orbit tracking routine. This is
>relatively straightforward given accurate orbital elements, time and
>station location.

Sounds quite involved - fortunate that you have such a nice closed-form predition method. I was wondering; are there any residual effects to deal with due to spin motion of the satellite? some higher than doppler phasing effects or does the circular polarized antennas take care of it?

Re: Doppler tracking on HF:
I'm looking into following reasonably fast-changing doppler effects, i.e., a fraction of a symbol time. The method that I'm interested in for estimating doppler comes from LPC methodology and the ideas are well developed - extracting pitch, or in other words specifying a filter from the signal. I'm not sure how well it will work for what I am doing, but I'm willing to give it a chance.

Anyway, the modem will have to go on the back burner for a while - something else has come up that needs to be done.

--Johan

72. Re: [HFSIG:1230] Re: BTL performance tests

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Wed, 19 Jun 1996 09:18:16 -0800
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19960619171816.0d271ab0@ucs.orst.edu>
Rendered body type: text/plain

Hi Rob,

>Ultimatly I would like to use BTL as a platform for more complex modes
>such as PACTOR II or Clover II or other multicarrier modes that may be
>developed through HFSIG. I'm also keen to implement very low SNR modes,
>such as DBPSK with heaps of ECC using a type II hybrid ARQ protocol.
>
>I intend to release up to version 2 as public domain freeware.
>

>
>Cheers,
>
>Rob Glassey, G0VTQ
>
>

Excellent!

Would you consider uploading a program version to HFSIG? If you do, also
please provide a ".txt" version for the folks that browse the files with
their web browser.

Keep us posted on your progress - sounds like good stuff.

--Johan

73. Re: [HFSIG:1235] Re: Further thoughts on pulse shaping for HF

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 19 Jun 1996 16:26:49 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199606192326.QAA23790@servo.qualcomm.com>
Rendered body type: text/plain

>Sounds quite involved - fortunate that you have such a nice closed-form
>predition method. I was wondering; are there any residual effects to deal
>with due to spin motion of the satellite? some higher than doppler phasing
>effects or does the circular polarized antennas take care of it?

As a matter of fact, yes. James Miller has pointed out that the S-band
antenna on AO-13 is not mounted along the center of mass, so if the
antennas are not lined up on earth the antenna moves to and fro along
the line of sight as the spacecraft spins. This introduces a
sinusoidal doppler shift on the order of 5 Hz, which is significant to
me.

Of course, this may be somewhat academic as I probably won't get this
stuff on mode S before AO-13 burns up, and P3D is 3-axis stabilized.

Phil

74. (no subject)

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Thu, 20 Jun 1996 13:28:28 +0200
To: hfsig@tapr.org
Message-ID: <s1c953b2.040@KIDD.CO.ZA>
Rendered body type: text/plain

If you have a sound blaster your best bet (and cheapest) would be to record the satilite on disk the play it back to
the DSP. 10 minutes will take about 5MB. You may need make your DSP code more independent of the PC (buffer output etc) or get another PC!

Its actually quite simple to write a C program to play the sound file and act as a dumb terminal at the same time.
==========
I see. Yes I do have another PC. BTW what is the bandwidth of the DAT tape ? I suppose it has now phase distorsion ie wow & flutter ?

danie

75. Re: [HFSIG:1238]

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Thu, 20 Jun 1996 11:37:50 -0400
To: hfsig@tapr.org
Message-ID: <1C971760.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

I'm curious, what would this C program look like?

     73s de
     Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1238]
Author: hfsig@tapr.org at BALT.SMTP
Date: 6/20/96 6:36 AM

```
If you have a sound blaster your best bet (and cheapest) would be to record the
satilite on disk the play it back to
the DSP. 10 minutes will take about 5MB. You may need make your DSP code more
independent of the
PC (buffer output etc) or get another PC!

Its actually quite simple to write a C program to play the sound file and act as
a dumb terminal at the same time.
============================
I see. Yes I do have another PC. BTW what is the bandwidth of the DAT tape ? I
suppose it has now phase
distorsion ie wow & flutter ?

danie
```

76. [HFSIG:1239] Re:

```
From: Robert.Glassey@nmp.nokia.com
Date: Thu, 20 Jun 96 18:36:59 +0100
To: hfsig@tapr.org, LANIER.R.A-@smtpgty.bwi.wec.com
Message-ID: <H000029201e4bf8e@MHS>
Rendered body type: text/plain
```

```
>> Its actually quite simple to write a C program to play the sound file
>> and act as a dumb terminal at the same time.

>      I'm curious, what would this C program look like?
>
>      73s de
>      Tony,  KE4ATO

There's libraries available on simtel for the SB stuff. It would look
something like this:

main()
{

soundblaster_reset(); /* also checks for card */

buffer_mem=faralloc(128*1024l); /* 64 k buffer when page aligned */

buffer=[page alignment of buffer_mem pointer ie to get XX00:0000]
/* this is so the DMA doesn't cross a 64K page boundry */

[setup interupt on SB IRQ]

[set DMA parameters - port commands]

[set sound blaster sample rate -port commands]

file=fopen("sat_rec","rb");

[read 32k from file into buffer]

[start SB playing in auto init mode]

quit=0;
buffer_flag=0;
while(!quit)
  {
/* double buffered disk read for SB */
  dma=[read DMA pointer]
  if (dma > 16384) && (buffer_flag==0)
    {
    /* if was lo and now playing hi half, load new low half*/
    [read next 16k to low half of buffer]; buffer_flag=1;
    }
  if (dma < 16384) && (buffer_flag==1)
    {
    /* if was hi and now playing lo half, load new high half*/
    [read next 16k to high half of buffer]; buffer_flag=0;
    }

/* very dumb terminal */
  if(kbhit())
    {
    ch=getch();
```

```
    if(ch==27) quit=1; /* esc quits */
    [ output ch to com port ]
    }
  [if char available at com port] putc(ch);
  }

stop_sb();
remove_sb_interupt();
fclose(file)
farfree(buffer_mem)
}

sb_interupt()
{
[acknoledge SB interupt - sb port commands]
out 20h,20h
}

I'm bound to have forgotten something, and have brushed over many things
but this is roughly how it would look. There's lots of other error and
end of file stuff as well.

To record the sample there's a few simple but good programs on simtel
which will do the trick nicely. Actually there might even be a
background TSR program that will play the soundfile in the background so

you can just run your normal terminal program. Or maybe use a standard
windows player and terminal, that might be even easier!

Cheers,

Rob
```

77. info on C31 DSK ?

```
From: zs6awk@global.co.za (Danie Brynard)
Date: Thu, 20 Jun 1996 22:02:32 +0200
To: hfsig@tapr.org
Message-ID: <199606210001.WAA06699@lin01.global.co.za>
Rendered body type: text/plain

Does anybody have more info on the new TI TMS320C31 DSK ? Like amount of RAM
and release date ? How good is the debugger ie compared to say 56002 OnCE ?

danie zs6awk
```

78. Re: [HFSIG:1239] Re:

```
From: Robert.Glassey@nmp.nokia.com
Date: Fri, 21 Jun 96 14:56:54 +0100
To: hfsig@tapr.org
Message-ID: <H000029201e58799@MHS>
Rendered body type: text/plain

>      Thanks for the code Rob, I appreciate it. Can this code be
>      compiled under Linux using GCC+?
>
>      Tony,  KE4ATO

Of course its only pseduo code, but it is intended for a PC under DOS.
Under Linux, you'd be better off using the sound card drivers that
already exist for Linux. Sorry I don't know very much about these, only
that they exist.

PS. I've just notices an error in my pseudo code, I allocated a 64
buffer then only used 32K of it (16K for each half of the bouble buffer)
You only really needed to allocate 32K thus:


buffer_mem=faralloc(64*1024l); /* 32 k buffer when page aligned */

buffer=[page alignment of buffer_mem pointer ie to get XX00:0000]
/* this is so the DMA doesn't cross a 64K page boundry */

This defines a page aligned 32K buffer in an alocated block of 64K.

It would have worked before, it just allocated twice as much memory as
required.

Cheers,
```

Rob

79. Question on orthoganal tone spacing

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Thu, 27 Jun 1996 09:32:28 -0800
To: HFSIG@tapr.org
Message-ID: <1.5.4.16.19960627173228.344f05c6@ucs.orst.edu>
Rendered body type: text/plain

Hi folks,

HFSIG been quiet lately?

I have a question that I would appreciate some insight - I'm a bit confused.

My multi-channel modulation scheme uses four parallel tones, each modulated at a rate of 75 symbols per second (bauds). If I understood our earlier discussion, each tone should fit into a 2*75=150 Hz slot. Why does this not follow the orthogonal 1/T rule for orthogonal spacing, that would put tones at 75 Hz spacing, not 150 as above?

To check this out, I looked at the MIL-STD-188 16-tone specifications which is quite similar, i.e., 75 baud with 110 Hz spacing. I tried this spacing but experience a substantial amount of inter-channel interference. Demodulation still works but with a serious performance hit - the recovered channel constellations are really smeared. However, when I go to the 150 (i.e. 2*T) spacing - the constallations are text-book quality.

So - what am I missing ?

---------------FYI---
A little more of where this is heading: four 75 baud tones with DQPSK modulation gives 600 bps. Various forms of diversity is possible. The highest form of diversity is when each channel carries the same information in DBPSK, in which case the rate is only 75 bps, but at four times diversity.

It would be possible to do what the Pactor-II folks do and up the DQPSK (4-DPSK) to 8-DQPSK that would essentially yield 1200 bps. Again, it is possible to use various forms of diversity. For example, using 8-DQPSK, and a factor of two diversity, would yield 600 bps.

The idea is to use an interleaved convolutional code, something along the lines that Phil have been working on. The waveform is 600 Hz wide at the -50 dB level, and could possibly be squeezed be into the 500 Hz constraint. However, I do not recommend folks getting ideas of using narrow crystal filters - severe phase distortion at the filter edges.

If time allows, I hope to do a show and tell demo of the project at the HFSIG meeting during the upcoming DCC (September 20-22, Seattle).

I sincerely hope that this effort will lead to a future open-architecture HF-digital protocol. All specifications will be freely available for those to implement it. Commercial implementations would be encouraged and I would be happy to offer consulting services if anyone is interested. HFSIG is there to encourage development and exchange ideas.

--Johan, KC7WW

80. Help communicate/translate in Italian

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Thu, 27 Jun 1996 11:57:49 -0800
To: HFSIG@tapr.org
Message-ID: <1.5.4.16.19960627195749.0d3f5f2c@ucs.orst.edu>
Rendered body type: text/plain

Hi,

Is there a kind soul out there that can help us communicate with a very keen and interested Italian gentleman?

I have been contacted several times by this keen individual, and although I can help myself in several languages, Italian unfortunately is not one of them - I hate to dissapoint, but I need a bit more than a few hand-waving gestures explaining how to get started using a DSP modem.

The gentleman is located in Milan. Any help would be greatly appreciated.

Thanks a lot.

--Johan

81. Re: [HFSIG:1243] Question on orthoganal tone spacing

From: Barry McLarnon VE3JF <bm@lynx.ve3jf.ampr.org>
Date: Fri, 28 Jun 1996 15:31:17 +0000 (GMT)
To: hfsig@tapr.org
Message-ID: <199606281531.PAA01611@lynx.ve3jf.ampr.org>
Rendered body type: text/plain

> I have a question that I would appreciate some insight - I'm a bit confused.
>
> My multi-channel modulation scheme uses four parallel tones, each modulated
> at a rate of 75 symbols per second (bauds). If I understood our earlier
> discussion, each tone should fit into a 2*75=150 Hz slot. Why does this not
> follow the orthogonal 1/T rule for orthogonal spacing, that would put tones
> at 75 Hz spacing, not 150 as above?

The 1/T rule does apply here - it *should* work with 75 Hz spacing.

> To check this out, I looked at the MIL-STD-188 16-tone specifications which
> is quite similar, i.e., 75 baud with 110 Hz spacing. I tried this spacing
> but experience a substantial amount of inter-channel interference.
> Demodulation still works but with a serious performance hit - the recovered
> channel constellations are really smeared. However, when I go to the 150
> (i.e. 2*T) spacing - the constallations are text-book quality.
>
> So - what am I missing ?

In the case of MIL-STD-188, there is a "guard interval" of about 4.2 ms
at the beginning of the 13.3 ms symbol interval to reduce ISI caused
by multipath. The integrate-and-dump (or equivalent) doesn't start
until after the guard interval, so T = 9.1 ms and 1/T = 110 Hz.

If your scheme isn't working at 1/T spacing, it seems like an
implementation issue. Perhaps you're using some kind of baseband pulse
shaping which is destroying the orthogonality of the tones at minimum
spacing?

Barry

--

  Barry McLarnon VE3JF/VA3TCP | Internet: bm@hydra.carleton.ca
  Ottawa Amateur Radio Club | AMPRnet: bm@lynx.ve3jf.ampr.org
  Packet Working Group | Web: http://hydra.carleton.ca

82. Re: [HFSIG:1245] Re: Question on orthoganal tone spacing

From: Johan Forrer <forrerj@ucs.orst.edu>
Date: Fri, 28 Jun 1996 20:32:59 -0800
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19960629043259.3667fa2c@ucs.orst.edu>
Rendered body type: text/plain

Hi Barry,

>
>The 1/T rule does apply here - it *should* work with 75 Hz spacing.
>
>> To check this out, I looked at the MIL-STD-188 16-tone specifications which
>> is quite similar, i.e., 75 baud with 110 Hz spacing. I tried this spacing
>> but experience a substantial amount of inter-channel interference.
>> Demodulation still works but with a serious performance hit - the recovered
>> channel constellations are really smeared. However, when I go to the 150
>> (i.e. 2*T) spacing - the constallations are text-book quality.
>>
>> So - what am I missing ?
>
>In the case of MIL-STD-188, there is a "guard interval" of about 4.2 ms
>at the beginning of the 13.3 ms symbol interval to reduce ISI caused
>by multipath. The integrate-and-dump (or equivalent) doesn't start
>until after the guard interval, so T = 9.1 ms and 1/T = 110 Hz.
>
>If your scheme isn't working at 1/T spacing, it seems like an
>implementation issue. Perhaps you're using some kind of baseband pulse
>shaping which is destroying the orthogonality of the tones at minimum
>spacing?

>
>Barry
>
>--
> Barry McLarnon VE3JF/VA3TCP | Internet: bm@hydra.carleton.ca
> Ottawa Amateur Radio Club | AMPRnet: bm@lynx.ve3jf.ampr.org
> Packet Working Group | Web: http://hydra.carleton.ca
>
>

Thanks for the response - you make a good point. It may well be an
implementation issue as there are many factors involved.

So ... I did a bit of research to understand things a bit better:

I created a signal like the MIL-STD-188 16-tone.
The way that this is done is to modulate all the carriers, sum them in one
signal and then time-domain filter the ensamble using a pulse shape that has
a "rounded" edges. The edges of this pulse shape are made up of 1/2 Hamming
rolloff cones that has a duration, the length of the guard zone (2.12 ms).
The main portion of the pulse top then is at unity.

The resultant signal audibly sounds reasonable, none of those ugly clicking
noises, a kind of a warbling-buzzing sound. My spectrum analyzer showed the
carriers are contained as expected and the noise floor is some 30-40 dB down
from the tone carriers.

Result: yes, indeed, the demodulator demodulates the individual signals OK,
but the recovered constellations are *not* as clean as what I would like to
see. Maybe some more implementation issues that may make some improvement?

I compared this to my individually-RC pulse-shaped signal waveform. Note
that here I use 2*T channel spacing, which is just slighty more than the
MIL-STD-188 tone spacing
(450 vs 440 Hz). I noted significant differences; The dynamic range between
constellation points are at least an order of magnitude better, and the
spurious levels in the spectrum are at least twice a low.

Thus I suspect that the RC pulse shaping has brought about a cleaner
emission and also improved the eye opening substantially. I would be as
brave as to say that I have high hopes that the scheme may be usable at 8-DQPSK.

OK so where does this leave us with the issue of orthogonality?

I do suspect that it may be practically impossible to achieve that, i.e.,
having that accuracy between the tone set. It seem to take only a very small
error to distroy the orthogonality to the point where it becomes nearly
useless. This may be why the MIL-STD-188 set also does not use the 1/T
spacing?. From your description above, the effectively space the tones is
110 Hz spacing (based on the 9.09 ms pulse). So we are already removed from
the 1/T orthogonal spacing.

Long story, thanks for your patience, but thought it was worth looking into.

73's

--Johan

83. Re: [HFSIG:1246] Re: Question on orthoganal tone spacing

From: Phil Karn Jr <karn@baa01075.slip.digex.net>
Date: Sat, 29 Jun 1996 01:25:41 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199606290525.BAA00279@baa01075.slip.digex.net>
Rendered body type: text/plain

>OK so where does this leave us with the issue of orthogonality?

Johan,

I'm out of town so I don't have access to my textbooks, but I *think*
the issue is whether you're using coherent or noncoherent detection.
With a coherent carrier reference at the receiver, you can use carrier
spacing twice as dense as with noncoherent detection.

Since you are using DQPSK, I assume you're doing noncoherent detection.
That could explain your need for greater tone spacing.

Phil

84. Re: [HFSIG:1247] Re: Question on orthoganal tone spacing

From: Barry McLarnon VE3JF <bm@lynx.ve3jf.ampr.org>
Date: Sat, 29 Jun 1996 16:34:04 +0000 (GMT)
To: hfsig@tapr.org
Message-ID: <199606291634.QAA02843@lynx.ve3jf.ampr.org>
Rendered body type: text/plain

> >OK so where does this leave us with the issue of orthogonality?
>
> Johan,
>
> I'm out of town so I don't have access to my textbooks, but I *think*
> the issue is whether you're using coherent or noncoherent detection.
> With a coherent carrier reference at the receiver, you can use carrier
> spacing twice as dense as with noncoherent detection.
>
> Since you are using DQPSK, I assume you're doing noncoherent detection.
> That could explain your need for greater tone spacing.
>
> Phil

This is true, but with coherent detection you can achieve a minimum orthogonal carrier spacing of 1/(2T). With noncoherent detection, the minimum spacing is 1/T as previously stated. A good example of a practical system using 1/T carrier spacing and noncoherent detection (DQPSK) is the Eureka 147 Digital Audio Broadcast system. For Mode 2 DAB, which we use here in Canada at L Band, the parameters are:

Transmitted symbol length T = 312.5 us, composed of:
Tg = 62.5 us (guard interval)
Ts = 250 us (symbol length in the demodulator)

Carrier spacing = 1/Ts = 4 KHz

There are 384 carriers, producing an overall bandwidth of 1.536 MHz.

I suspect that Johan's modem may not be performing well with 1/T carrier spacing because he is using windowing which destroys the orthogonality. Have you tried using rectangular windowing, Johan? And, of course, if you implement a guard time to avoid ISI, you must adjust the carrier spacing so that the orthogonality is maintained in the demodulator (spacing of 1/Ts rather than 1/T).

Barry

--
Barry McLarnon VE3JF/VA3TCP | Internet: bm@hydra.carleton.ca
Ottawa Amateur Radio Club | AMPRnet: bm@lynx.ve3jf.ampr.org
Packet Working Group | Web: http://hydra.carleton.ca

July 1996

Original source file: hfsig.9607
Source: https://archive.org/details/tapr-1997
Recovered from: https://archive.org/download/TAPR-HF_1996-07/TAPR-HF_1996-07.txt

1. About a Digital Comms Theory Book

From: Paul Sadowski <sadowski@cris.com>
Date: Mon, 1 Jul 1996 21:38:19 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <2.2.16.19960701214156.36d71956@pop3.cris.com>
Rendered body type: text/plain

Request Please

I seem to recall there was a mention of a book on Digital Comms that
was due out "real soon"... somebody here I think was involved in it
being written and maybe TAPR was going to publish / carry it??? I
even think there was a draft table of contents mentioned...

Anyone know what the story is ???

Mahalo and ALOHA

2. Re: [HFSIG:1249] About a Digital Comms Theory Book

From: Greg Jones <wd5ivd@tapr.org>
Date: Mon, 1 Jul 1996 21:15:00 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199607020215.VAA02178@tapr.org>
Rendered body type: text/plain

There is a link to it on www.tapr.org on the bottom of the page with the
details.

Tom and myself are meeting the 4th to complete his edits. Then it is one
last run through by the proof reader and then off to the press. We are
shooting to go to the printers by the middle of July and then the date after
that will be set by them on when we get them -- figure 4-5 weeks+ depending
on backlog at their house.

Still no price, since we don't have a quote yet from e printers.

Cheers - Greg

>
> Request Please
>
> I seem to recall there was a mention of a book on Digital Comms that
> was due out "real soon"... somebody here I think was involved in it
> being written and maybe TAPR was going to publish / carry it??? I
> even think there was a draft table of contents mentioned...
>
> Anyone know what the story is ???
>
>
> Mahalo and ALOHA
>
>
>

3. PC-DSP and PC-SIM for Windows - TAPR Group Purchase

From: Greg Jones <wd5ivd@tapr.org>
Date: Tue, 2 Jul 1996 00:08:39 -0500 (CDT)
To: tapr-bb@tapr.org (TAPR-BB mailing), hfsig@tapr.org (HF SIG mailing),
        dsp-93@tapr.org (DSP-93 Build)
Cc: psr@tapr.org (PSR)
Message-ID: <199607020508.AAA14429@tapr.org>
Rendered body type: text/plain

PC-DSP and PC-SIM for Windows - TAPR Group Purchase

For the past several months, the subject of digital signal processing has
been discussed on the TAPR DSP-93 and HFSIG e-mail lists. Software designed
to facilitate the further learning and modeling of DSP entities has also
been discussed (PC-DSP). Mention was made of a DSP course utilizing a text
and a program called PC-DSP. The text is aptly named "Digital Signal

Processing-A Laboratory Approach Using PC-DSP" by Oktay Alkin, PhD.

Jim Kauten, KO4RQ, checked with PC Solutions, the developer of the software, and it was discovered that a Windows version is available. There are versions available for Win 3.1x, Win NT, and Win 95.

After discussion with PC Solutions, they are willing to give a volume discount if there is enough interest. Jim announced the offer several weeks ago and approximately 30 people expressed an interest. Based on this interest TAPR will do a group purchase.

The cost per unit for either the PC-SIM or PC-DSP (for Windows) package will be $220.00 US plus $6 shipping/handling (*). The normal price is $256 + s/h from PC Solutions. A savings of $30+ for all those involved in the group purchase.

21 orders must be placed with Dorothy at the TAPR office before the purchase will be made. These are orders (i.e. check, MO, or Visa/MC). This is not a call to generate a list that will be contacted at some future time. As with past group purchases, monies collected for the purchase will not be deposited until the order is placed.

This information is available at:
http://www.tapr.org/tapr/html/dsp93.html

Overview:

The PC-DSP is an interactive, menu-driven software package used for: waveform synthesis using a variety of methods, basic signal operation, fast Fourier transforms, convolution and correlation, solution of difference equations, analysis and design of IIR and FIR filters, digital filter simulation and code generation, and power spectrum estimation using classical and modern techniques. Some key features of PC-DSP listed include: GNUPLOT support, code generation, macro compiler, dialog compiler, sound file support, data file formats, and compatibility with PC-SIM.

PC-SIM is described as a continuous- and discrete-time simulator that is used for time-domain simulation of systems described by block diagrams. It was designed to be a flexible and open-ended tool to allow simulation of a broad range of systems encountered in communications, signal processing, and control theory. Some of the key features mentioned include: pre-defined components, code generation, sound file support, and compatibility with PC-DSP.

Demo versions of both programs are available from the PC Solutions web site: http://www.dspsolutions.com.

Information regarding the software should be directed to Jim Kauten, MD, KO4RQ (kauten@mindspring.com).

Orders for software should be directed to the TAPR office tapr@tapr.org

TAPR would like to thank Jim, KO4RQ, for his effort in organizing this purchase.

TAPR
8987-309 E. Tanque Verde Rd., #337
Tucson, AZ 85749-9399

Office: 817-383-0000
Fax: 817-566-2544
E-mail: TAPR@TAPR.ORG

Office Hours: Tue - Fri, 9am - 12noon & 3pm - 5pm Central Time

(*) Note: There will be no 10% membership discount on this purchase.

July 2nd, 1996

4. Re: [HFSIG:1251] PC-DSP + PC-SIM

From: chbrain@dircon.co.uk (Charles Brain)
Date: Tue, 02 Jul 1996 06:58:27 +0100
To: hfsig@tapr.org
Message-ID: <199607020606.AA16754@felix.dircon.co.uk>
Rendered body type: text/plain

Hello All,

>simulation and code generation, and power spectrum estimation using
>classical and modern techniques. Some key features of PC-DSP listed
>include: GNUPLOT support, code generation, macro compiler, dialog compiler,

I already have this software and I think it is very good. I am currently
using PC-SIM to model a simple DS SS system.

Does anyone know where to get a hold of GNUPLOT, I have searched
the COMPUSERVE archives and all I can find is the manual !

Regards Charles G4GUO

---

5. Re: [HFSIG:1252] Re: PC-DSP + PC-SIM

From: Frode Weierud <frode@dxcern.cern.ch>
Date: Tue, 2 Jul 1996 09:21:47 +0200 (MET DST)
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <Pine.ULT.3.91.960702091539.17640A-100000@dxcern.cern.ch>
Rendered body type: text/plain

On Tue, 2 Jul 1996, Charles Brain wrote:

> Does anyone know where to get a hold of GNUPLOT, I have searched
> the COMPUSERVE archives and all I can find is the manual !
>
> Regards Charles G4GUO
>
>

Here are a few excerpts from the Gnuplot FAQ telling how to get hold of
the FAQ and where to find the source code. The FAQ has answers to many
other questions as well so it is advisable to get hold of that first.

```
    Q0.1: Where do I get this document?
          This document is posted about once every two weeks to the
          newsgroups comp.graphics.apps.gnuplot, comp.answers and
          news.answers. Like many other FAQ's, its newest (plaintext)
          version is available via anonymous ftp from
          ftp://rtfm.mit.edu/pub/usenet/news.answers/graphics/gnuplot-fa
          q.

          If you have access to the WWW, you can get the newest version
          of this document from
          http://www.uni-karlsruhe.de/~ig25/gnuplot-faq/

  Section 2: Setting it up

    Q2.1: What is the current version of gnuplot?
          The current version of gnuplot is 3.5, which is a bugfix
          release over 3.4.

          Version 3.6 is in beta status. Please note that this is still
          unstable, and may not compile correctly on your system.

   Q2.2: Where can I get gnuplot?
          All of the later addresses refer to ftp sites. Please note that
          it is preferable for you to use the symbolic name, rather than
          the IP address given in brackets, because that address is much
          more subject to change.

          The official distribution site for the gnuplot source is
          ftp.dartmouth.edu [129.170.16.4, soon to be 129.170.8.11], the
          file is called /pub/gnuplot/gnuplot3.5.tar.Z. Official mirrors
          of that distribution are (for Australia) ftp.monash.edu.au
          [130.194.11.18] and (for Europe) irisa.irisa.fr
          [131.254.254.2]. You can also get it from your friendly
          neighbourhood comp.sources.misc archive.
```

MS-DOS and MS-Windows binaries are available from

+ oak.oakland.edu (North America) [141.210.10.117] as
  /Simtel/msdos/plot/gpt35*.zip,
+ garbo.uwasa.fi (Europe) [128.214.87.1] as /pc/plot/gpt35*.zip
  and
+ archie.au (Australia) [139.130.4.6] as
  micros/pc/oak/plot/gpt35*.zip.

The files are: gpt35doc.zip, gpt35exe.zip, gpt35src.zip and
gpt35win.zip.

There is a special MS-DOS version for 386 or better processors;
it is available from the official gnuplot sites as DOS34.zip.

OS/2 2.x binaries are at ftp-os2.nmsu.edu [128.123.35.151], in
/os2/2.x/unix/gnuplt35.zip.

Amiga sources and binaries are available from ftp.wustl.edu
[128.252.135.4] as /pub/aminet/util/gnu/gnuplot-3.5.lha; there
are numerous mirrors of this distribution, for example
ftp.uni-kl.de, oes.orst.edu or ftp.luth.se.

The NeXTSTEP front end can be found at
ftp://next-ftp.peak.org/pub/next/binaries/plotting/ as
Gnuplot1.2_bin.tar.Z.

A version for OS-9/68K can be found at cabrales.cs.wisc.edu
[128.105.36.20] as /pub/OSK/GRAPHICS/gnuplot32x.tar.Z; it
includes both X-Windows and non - X-windows versions.

There is a version for the Macintosh at
ftp://ftp.ee.gatech.edu/pub/mac/gnuplot/ which includes
binaries for 68000-based Macs with and without FPU and native
support for PowerMacs.

Versions for the Atari ST and TT, which include some GEM
windowing support, are available from
ftp://ftp.uni-kl.de/pub/atari/graphics/, as gplt35st.zip and
gplt35tt.zip. They work best under MiNT.

Executable files, plus documentation in Japanese, exist for the
X680x0 on
ftp://ftp.csis.oita-u.ac.jp/pub/x68k/fj.binaries.x68000/vol2.

People without ftp access can use an ftp-mail server; send a
message saying 'help' to bitftp@pucc.bitnet (for BITNET only)
or to ftpmail@ftp.dartmouth.edu.

For a uuencoded copy of the the gnuplot sources (compressed tar
file), send this as the body of a message to
ftpmail@ftp.dartmouth.edu:

```
open
cd pub/gnuplot
mode binary
get gnuplot3.5.tar.Z
quit
```

If you have some problem, you might need to stick

```
reply-to  <your-email-address-here>
```

before all the above.

It is a good idea to look for a nearby ftp site when
downloading things. You can use archie for this. See if an
archie client is installed at your system (by simply typing
archie at the command prompt), or send mail to archie@sura.net
with the word 'help' in both the subject line and the body of
the mail. However, be aware that the version you find at a near
ftp site may well be out of date; check the last modification
date and the number of bytes against the newest release at one
of the official servers.

You can obtain a beta release of gnuplot 3.6 from
ftp://cmpc1.phys.soton.ac.uk/pub/.

73's Frode, F/LA2RL

Frode Weierud Phone : 41 22 7674794
CERN, SL, CH-1211 Geneva 23, Fax : 41 22 7679185
Switzerland E-mail : Frode.Weierud@cern.ch

6. PSK Question..

From: zs6awk@global.co.za (Danie Brynard)
Date: Tue, 02 Jul 1996 22:26:23 +0200
To: hfsig@tapr.org
Message-ID: <199607030024.WAA24050@lin01.global.co.za>
Rendered body type: text/plain

I am sorry if my questions are out of sync with the ongoing PSK/QPSK
discussions but could somebody just answer the following perhaps for me again:

1) A Costasloop for BPSK demodulation will typically lock 0 or 180 degrees
out of phase with the incoming carrier. Can it also lock 90 or 270 degrees
out of phase and what would typically be the reasons for this ?

2) What is the minimum signal to noise required for the following BPSK
demodulator techniques:

- squaring loop PLL type
- PSK Costas Loop
- or perhaps what is the minimum SNR for a PLL to maintain phase lock ?

Any references would of course be welcomed ie page number in a textbook. I
have implemented two RTTY modems for the DSP56002EVM. The one is based on a
DPLL for fsk demod. I was wondering what the minimum theoretical SNR is that
would keep the DPLLL in lock...from there my thoughts & questions.

Regards, Danie zs6awk
zs6awk@global.co.za

7. Re: [HFSIG:1254] PSK Question..

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)
Date: Tue, 2 Jul 96 16:30:46 CDT
To: hfsig@tapr.org
Message-ID: <9607022130.AA07929@eagle.aud.alcatel.com>
Rendered body type: text/plain

> I am sorry if my questions are out of sync with the ongoing PSK/QPSK
> discussions but could somebody just answer the following perhaps for me again:
>
> 1) A Costasloop for BPSK demodulation will typically lock 0 or 180 degrees
> out of phase with the incoming carrier. Can it also lock 90 or 270 degrees
> out of phase and what would typically be the reasons for this ?

Danie: there is in fact zero error voltage at the wrong lock
points (90, 270). However, the loop-sense is backwards at these two
points. Thus the loop will not stay locked at the wrong points as
drift, noise, etc. will cause the loop to 'kick-out' of those two
points. The feedback loop is only stable at the 0 and 180 degree
points.

>
> 2) What is the minimum signal to noise required for the following BPSK
> demodulator techniques:
>
> - squaring loop PLL type
> - PSK Costas Loop
> - or perhaps what is the minimum SNR for a PLL to maintain phase lock ?

Danie: This is a difficult question to answer analytically, and I
think empirical results are more trustworthy. I think that the answer
is implementation dependent. No good references to give you right now.

- Tom

Tom McDermott
mcdermot@aud.alcatel.com

8. Re: [HFSIG:1255] Re: PSK Question..

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 2 Jul 1996 16:02:09 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607022302.QAA13658@servo.qualcomm.com>
Rendered body type: text/plain

> 2) What is the minimum signal to noise required for the following BPSK
> demodulator techniques:
>
> - squaring loop PLL type
> - PSK Costas Loop
> - or perhaps what is the minimum SNR for a PLL to maintain phase lock ?

The squaring loop and Costas loop behave identically with appropriate
choice of filters. There's no theoretical performance advantage of one
over the other. The main differences are practical, such as the lower
frequencies inherent in the Costas loop approach (which makes it more
easily implemented with DSP).

When designed for BPSK, both loops behave about 6 dB worse than a CW
PLL (i.e., an ordinary PLL without any means for removing PSK
modulation). When designed for QPSK, they're both about 12 dB
worse. This is for relatively strong signals that are above a certain
threshold. Even worse, there is an additional "squaring loss" that
depends on the signal-to-noise ratio in the loop that increases the
loss at low SNR. This behavior is inherent in a loop that removes
modulation -- it's not the fault of either the Costas or squaring loop
design.

Phil

9. Update on PC-DSP and PC-SIM Group Purchase

From: Greg Jones <wd5ivd@tapr.org>
Date: Tue, 2 Jul 1996 23:00:17 -0500 (CDT)
To: dsp-93@tapr.org (DSP-93 Build), hfsig@tapr.org (HF SIG mailing),
tapr-bb@tapr.org (TAPR-BB mailing), psr@tapr.org (PSR)
Message-ID: <199607030400.XAA15843@tapr.org>
Rendered body type: text/plain

Update on TAPR Group Purchase of the PC-DSP and PC-SIM software. It was
unclear in the first posting that this was indeed a software bundle. Sorry
for any problems this might have caused.

We received 5 orders just today at the TAPR office -- so get those orders
rolling!

PC-DSP and PC-SIM for Windows (bundle) - TAPR Group Purchase

For the past several months, the subject of digital signal processing has
been discussed on the TAPR DSP-93 and HFSIG e-mail lists. Software designed
to facilitate the further learning and modeling of DSP entities has also
been discussed (PC-DSP). Mention was made of a DSP course utilizing a text
and a program called PC-DSP. The text is aptly named "Digital Signal
Processing-A Laboratory Approach Using PC-DSP" by Oktay Alkin, PhD.

Jim Kauten, KO4RQ, checked with PC Solutions, the developer of the software,
and it was discovered that a Windows version is available. There are
versions available for Win 3.1x, Win NT, and Win 95.

After discussion with PC Solutions, they are willing to give a volume
discount if there is enough interest. Jim announced the offer several weeks
ago and approximately 30 people expressed an interest. Based on this
interest TAPR will do a group purchase.

The cost for the software bundle (PC-SIM and PC-DSP (for Windows)) will be
$220.00 US plus $6 shipping/handling (*). The normal price is $256 with s/h
from PC Solutions. A savings of $30 for all those involved in the group
purchase.

21 orders must be placed with Dorothy at the TAPR office before the
purchase will be made. These are orders (i.e. check, MO, or Visa/MC). This
is not a call to generate a list that will be contacted at some future time.
As with past group purchases, monies collected for the purchase will not be
deposited until the order is placed.

This information is available at:
http://www.tapr.org/tapr/html/dsp93.html

Overview:

The PC-DSP is an interactive, menu-driven software package used for:
waveform synthesis using a variety of methods, basic signal operation, fast
Fourier transforms, convolution and correlation, solution of difference
equations, analysis and design of IIR and FIR filters, digital filter
simulation and code generation, and power spectrum estimation using
classical and modern techniques. Some key features of PC-DSP listed
include: GNUPLOT support, code generation, macro compiler, dialog compiler,
sound file support, data file formats, and compatibility with PC-SIM.

PC-SIM is described as a continuous- and discrete-time simulator that is
used for time-domain simulation of systems described by block diagrams. It
was designed to be a flexible and open-ended tool to allow simulation of a
broad range of systems encountered in communications, signal processing, and
control theory. Some of the key features mentioned include: pre-defined
components, code generation, sound file support, and compatibility with
PC-DSP.

Demo versions of both programs are available from the PC Solutions web site:
http://www.dspsolutions.com.

Information regarding the software should be directed to Jim Kauten, MD,
KO4RQ (kauten@mindspring.com).

Orders for software should be directed to the TAPR office tapr@tapr.org

TAPR would like to thank Jim, KO4RQ, for his effort in organizing this
purchase.

TAPR
8987-309 E. Tanque Verde Rd., #337
Tucson, AZ 85749-9399

Office: 817-383-0000
Fax: 817-566-2544
E-mail: TAPR@TAPR.ORG

Office Hours: Tue - Fri, 9am - 12noon & 3pm - 5pm Central Time

(*) Note: There will be no 10% membership discount on this purchase.

July 2nd, 1996

10. One more time

From: Greg Jones <wd5ivd@tapr.org>
Date: Tue, 2 Jul 1996 23:21:00 -0500 (CDT)
To: dsp-93@tapr.org (DSP-93 Build), hfsig@tapr.org (HF SIG mailing)
Message-ID: <199607030421.XAA17015@tapr.org>
Rendered body type: text/plain

Just in case the last update was a little unclear -- this should be real
clear for everyone :-)

Greg

The cost of the PC-SIM and PC-DSP for Windows package (includes both
programs) will be $220.00 US (*). Shipping and handling will be an
additional $6 for US deliveries. The standard non-discounted price for the
package is $256.00, inclusive of s/h, from PC Solutions. The TAPR group
purchase plan nets a savings of $30.00 for each person involved in the
purchase of the software package.

21 orders must be placed with Dorothy at the TAPR office before the
purchase will be made. These are orders (i.e. check, MO, or Visa/MC). This
is not a call to generate a list that will be contacted at some future time.

As with past group purchases, monies collected for the purchase will not be
deposited until the order is placed.

This information is available at:
     http://www.tapr.org/tapr/html/dsp93.html

11. Re: [HFSIG:1256] Re: PSK Question..

From: zs6awk@global.co.za (Danie Brynard)
Date: Wed, 03 Jul 1996 21:58:47 +0200
To: hfsig@tapr.org
Message-ID: <199607032357.VAA05539@lin01.global.co.za>
Rendered body type: text/plain

Thanks for the answers I received on my PSK question. However one thing I am
still not certain about: what is the minimum SNR required for a theoretical
ideal PLL to lock onto a CW signal ? (ok there is normally a pull-in range
and a hold-in range involved which might require different minimum SNR's)

Somebody wispered 6dB into my ear :-)

73 danie
zs6awk@global.co.za

12. Re: [HFSIG:1249] About a Digital Comms Theory Book

From: "Tony Godshall" <togo@of.net>
Date: Thu, 4 Jul 1996 12:22:56 +0000
To: hfsig@tapr.org
Message-ID: <199607042209.MAA02376@aloha.com>
Rendered body type: text/plain

[Paul Sadowski "[HFSIG:1249] About a Digital Comms " 1 Jul 96]

> Mahalo and ALOHA

Aloha! Are you in or from Hawaii? It would be good to get in touch
with others with digital HF interests. I'm in Pukalani Maui.

73 de WH6ZD

Best regards.
Anthony Godshall

-
Paradigm Shifting and Testing
Digital Models and Transformations
Project Engineer, Educator, Programmer

13. subscribe

From: John Fischer <john-f@bachrach.Com>
Date: Fri, 5 Jul 96 08:27:42 -0500
To: hfsig@tapr.org
Message-ID: <D75DD83101000000@bachrach.Com>
Rendered body type: text/plain

[Empty message body]

14. Re: TACO-2 low overhead protocol (was Re: Sound Card Interface)

From: "Anthony Godshall" <togo@of.net>
Date: Fri, 5 Jul 1996 09:44:40 +0000
To: hfsig@tapr.org
Message-ID: <199607051933.JAA16389@aloha.com>
Rendered body type: text/plain

[Sadowski,P, 7 Mar 96]

> US DoD went through extreme pain and agony over the overhead
> of the new protocols over low bandwidth channels. ... they came up with
> a low overhead protocol... with FEC option... called TACO 2
> (tactical comm protocol). I suggest the SIG look into this
> protocol.... DoD has been seeking IEEE or ISO standard for this
> protocol. Its tailor made for low BW RF links...

In case anyone else wants to follow up on the above, I found specs in
Zipped Wordperfect format at
  ftp://ftp.itsi.disa.mil/pub/library/nitfs_docs/taco2.zip
which were referenced from web page
  http://www.tasc.com/NITFS/NITFS_docs.html

Best regards.
Anthony Godshall

-
Paradigm Shifting and Testing
Digital Models and Transformations
Project Engineer, Educator, Programmer

15. HELP,(TNC)

From: Lu Hui-Hua <gzluhuih@public1.guangzhou.gd.cn>
Date: Sat, 06 Jul 1996 09:36:20 +0900
To: hfsig@tapr.org
Message-ID: <31DDB504.651C@public1.guangzhou.gd.cn>
Rendered body type: text/plain

hello everybody, here is BD7IX(Lu Hui-Hua) from P.R.CHINA.
somedays ago, i send a mail at here , said want to know how
to made a modem(tnc) myself, my plan is use TCM3105 or
AM7910(7911) chip to do it, but unfortunate, the chip is
not produce. soi think maybe motorola's MC145442 OR MC145443
can replace it. but mc145442(3) only can use 300 baud
(CCITT V.21 OR BELL 103).
who can tell me the alternative else?
and if i use MC145442(3), who can give the circuit design about it?
please tell me via e-mail bd7ix@amsat.org
thanks

Lu Hui-Hua*bd7ix*

16. FECC Chipsets (and an apology)

From: "Anthony Godshall" <togo@of.net>
Date: Fri, 5 Jul 1996 16:03:23 +0000
To: hfsig@tapr.org
Message-ID: <199607060152.PAA22253@aloha.com>
Rendered body type: text/plain

The apology is at the bottom. The question is, will anyone take me
seriously now?

Anyway, I found Phil's discussion of FEC very enlightening (Realaudio
and slides at http://www.tapr.org/tapr/html/taprstlouis.html#karn) -
it is well worth listenging carefully and rewinding some to get
though the background noise.

What I am wondering, with CD's so popular, is it possible to adapt
CD-ROM chipsets to radio? I searched the web and the newsgroups, and
the only ECC chip I could find mentioned was a single mention of a
L64715 BCH error correction chip from LSI Logic, and no comp.compression FAQ
(ftp://rtfm.mit.edu/pub/usenet/news.answers/compression-faq/)

The apology... I said...

> Aloha! Are you in or from Hawaii? It would be good to get in touch
> with others with digital HF interests. I'm in Pukalani Maui.

Oops. Color me newbie. Damn, I could have sworn I addressed that
properly. Anyway, I am sorry, as I wipe the egg off my face, for
wasting the bandwidth in such a juvenile way, and again now.

Best regards.
Anthony Godshall

-
Paradigm Shifting and Testing
Digital Models and Transformations
Project Engineer, Educator, Programmer

17. UNKN422.WAV

From: k6sti@n2.net (Brian Beezley)
Date: Sat, 6 Jul 1996 10:04:19 -0700
To: mwcook@cris.com
Cc: moon-net@vm.stlawu.edu, hfsig@tapr.org
Message-ID: <199607061704.KAA25622@ravel.n2.net>
Rendered body type: text/plain

Mike, I wanted to tell you about my ongoing efforts to crack your
UNKN422.WAV EME callsign mystery. I'm motivated primarily by the technical
challenge; the $100 prize you offer will barely pay for the electricity I've

expended! I think this is a really neat challenge you've come up with, so I'm copying this note to the EME reflector and to the TAPR HFSIG reflector (where there's recently been some discussion of CW decoding) in hopes of spurring others to try tackling the problem (perhaps one more time).

Earlier this week I seriously looked at the problem for the first time. I spent the better part of two days on it. Listening to the file without benefit of signal processing, I could just barely hear a tone pop out of the noise for a second or two. (The first time I listened, I heard only noise.) This was so unexpected that I downloaded the file again (http://www.webcom.com/af9y) to make sure I had the right one!

I cobbled together a special version of DSP Blaster to process the .WAV file instead of real-time, sound-card audio. I found neither its LMS noise reduction nor 15-wpm matched filter helpful.

For quite some time I've wondered whether a matched filter really is optimal for processing CW signals for detection by ear. A matched filter is optimal for threshold detection of signals in noise under certain conditions, but I wonder whether the conditions hold when preprocessing a signal for detection by ear. I don't know what the ear/brain does to detect CW, but it's certainly more complex than simple thresholding. For example, for many years I've preferred to use an SSB-bandwidth filter to recover weak HF CW signals rather than a 500-Hz CW filter. The additional noise somehow makes signal detection seem easier. (I haven't run a test to verify that it actually makes detection more reliable). I find listening through extremely narrow filters somewhat disconcerting because they shape the background-noise spectrum approximately to that of a sine wave at the filter center frequency. So your ear must then distinguish between the desired sine wave and weaker ones that bobble around in the background. Even though I estimate that DSP Blaster's 15-wpm matched filter improves CW detection by perhaps 1 dB for my ears, I sometimes prefer to detect CW in wideband noise (even though listening for long periods that way can be fatiguing). (It's interesting that most ops seem to vastly prefer narrowband CW filters.)

Since UNKN422 remained a mystery, I tried implementing a signal-processing technique I had planned for DSP Blaster version 2. I've had my Pentium-100 for one year and this was the first time I ever ran up against its processing-power limits. I had to recode the algorithm using Pentium-specific, floating-point optimizations (pipelining). It immediately yielded clearly audible dots and dashes throughout the .WAV file. Now I was getting somewhere! Right away I discovered I had misestimated the sending speed. I now was sure that recovering the callsign would be easy. I visualized the "K6STI Cracks UNKN422 with Secret Algorithm" headlines. So I listened carefully for some time, wrote down some possible calls, and consulted your list of 1000 EME callsigns, among which is the mystery callsign. Hmmmm...nothing matched very well. After more listening I found that I could mentally transform any given sound segment into my current best-guess callsign with very little wishful thinking! I was familiar with this phenomenon but thought I'd outgrown it years ago. I've heard 160-meter operators talk about it and I imagine it must be old-hat to you EME ops.

Twice I was quite sure of the callsign. I was ready to submit my winning entry and thought I'd take one last listen. Then I discovered that I heard completely different letters by altering playback speed and pitch! This was quite a shock to someone who prides himself on his hearing acuity. Although the dots and dashes are readily audible, the signal level is quite variable. (I assume this is due to some kind of moon multipath.) Often the code elements seem to have holes in them or to be smeared together. Sometimes it's not clear whether it's one long dash or a dot and dash. I also discovered that my callsign choices tended to depend critically on one or two crucial letters. It was very easy to persuade myself to ignore other sounds that really didn't fit but clearly weren't important because, after all, surely this was the correct callsign! The fascinating thing about this problem is that it involves both a technical and psychological challenge.

In frustration, I tried cascading a semi-matched filter with the processor-intensive technique. It didn't help. I really didn't want to guess at a callsign, so I kept listening over and over, tweaking algorithm constants. More new callsigns. I got so involved with the problem that my sleeping schedule became disrupted. When I reluctantly stopped listening to take a much-needed nap, I fitfully dreamed about weak CW! I decided it was time to return to normal life and finally gave up on the problem late one night. Temporarily.

I have one more technical trick to pull. In fact, although it's not implemented in the current version, this trick was the whole reason I started the DSP Blaster project. I plan to implement this technique in version 2 along with the processor-intensive algorithm. I think the

combination may provide enough S/N enhancement to yield a solution. It will take some experimentation to get right and a fair amount of programming to implement, so I expect it will be a little while in coming. Meanwhile, maybe someone else will decide to try a concerted attack on the problem.

I think your UNKN422.WAV challenge would be a really neat term project for university engineering students. It's a fascinating mystery, involves signal-processing theory, practical engineering, programming techniques, and human perception and psychology. It's really a lot of fun! You might try to locate an engineering professor somewhere who is also a ham and lay it on him.

73--Brian, K6STI
k6sti@n2.net

18. Re: [HFSIG:1263] HELP,(TNC)

From: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Date: Sat, 06 Jul 1996 19:51:19 -0400
To: bd7ix@amsat.org
Cc: hfsig@tapr.org
Message-ID: <199607062351.TAA02284@appstate-01.campus.mci.net>
Rendered body type: text/plain

At 08:44 PM 7/5/96 -0500, you wrote:
>hello everybody, here is BD7IX(Lu Hui-Hua) from P.R.CHINA.
>somedays ago, i send a mail at here , said want to know how
>to made a modem(tnc) myself, my plan is use TCM3105 or
>AM7910(7911) chip to do it, but unfortunate, the chip is
>not produce. soi think maybe motorola's MC145442 OR MC145443
>can replace it. but mc145442(3) only can use 300 baud
>(CCITT V.21 OR BELL 103).
>who can tell me the alternative else?
>and if i use MC145442(3), who can give the circuit design about it?
>please tell me via e-mail bd7ix@amsat.org
>thanks
>
>Lu Hui-Hua*bd7ix*
>
>
Lu Hui-Hua,

I also have been looking for the TCM 3105 without success. I talked to Texas Instruments and they have it marked as an obsolete part (either out of production or due to go out of production soon). TI directed me to:

MX-COM, Inc.
4800 Bethania Station Road
Winston-Salem, NC 27105-1201
1-800-638-5577
Fax: 910-744-5054

Their part, MX 469, 1200 & 2400 bps MSK Modem, is very similar to the TCM 3105. It has the Bell 202, but doesn't have the v23 feature. I have not studied the spec sheet closely, but it looks like the same circuit that was in the December 1995 QST, (p 39, IMP Modem), should work. The number of pins on the MX 469 is different than the TCM 3105 so the printed circuit board from FAR circuits would not work directly.

73's
Don, K4JPJ

19. Re: [HFSIG:1265] UNKN422.WAV

From: forrerj@peak.org (Johan Forrer)
Date: Sat, 06 Jul 1996 20:42:50 -0800
To: hfsig@tapr.org
Message-ID: <199607070339.UAA16114@PEAK.ORG>
Rendered body type: text/plain

Brian,

....

>For quite some time I've wondered whether a matched filter really is optimal
>for processing CW signals for detection by ear. A matched filter is optimal
>for threshold detection of signals in noise under certain conditions, but I
>wonder whether the conditions hold when preprocessing a signal for detection
>by ear. I don't know what the ear/brain does to detect CW, but it's
>certainly more complex than simple thresholding. For example, for many

>years I've preferred to use an SSB-bandwidth filter to recover weak HF CW
>signals rather than a 500-Hz CW filter. The additional noise somehow makes
>signal detection seem easier.

An interesting attempt.

I always understood that the theory of matched filters relates to a decision-making process that assumes Gaussian noise. Having had some discussions with weak signal microwave experts, I have been reminded that microwave signals pretty much are like dealing with Gaussian noise and matched filters are probably a reasonable approach. This, however, may perhaps not hold for CW/RTTY decoding on HF, where I suspect adjacent channel suppression and good filter skirts may be more important. There are a couple of good arguments along these lines made in Frerking's book, if you would like to follow up on that.

One other thing that you may give some thought; If the mystery CW is machine generated, it should have fairly accurate timing on the rising and falling edges. That then brings you to try sequency theory, i.e, doing FHT's (Fast Hademard Transforms) that may help recovering timing.

The moral of the story is, and if my understanding of the challenge is correct, is that the signals are below the level that commonly-known methods can handle. You will need to pull some clever tricks. Like for example using sequency theory, where you are using information that is spread out over time, not only from what happens within a short time window.

My 2 cents for what it's worth. Good luck with your efforts.

--Johan

20. vco block for simulink..

From: zs6awk@global.co.za (Danie Brynard)
Date: Sun, 07 Jul 1996 15:02:35 +0200
To: hfsig@tapr.org
Message-ID: <199607071700.PAA20481@lin01.global.co.za>
Rendered body type: text/plain

Does somebody know if simulink has a vco block and if not where can one get it ? I am considering to try a modem simulation on simulink but from what I hear there is no vco. (in the older versions in any way)

regards, danie

21. Re: [HFSIG:1256] Re: PSK Question..

From: zs6awk@global.co.za (Danie Brynard)
Date: Sun, 07 Jul 1996 15:43:01 +0200
To: hfsig@tapr.org
Message-ID: <199607071741.PAA22273@lin01.global.co.za>
Rendered body type: text/plain

>When designed for BPSK, both loops behave about 6 dB worse than a CW
>PLL (i.e., an ordinary PLL without any means for removing PSK
>modulation). When designed for QPSK, they're both about 12 dB
>worse. This is for relatively strong signals that are above a certain
>threshold. Even worse, there is an additional "squaring loss" that
>depends on the signal-to-noise ratio in the loop that increases the
>loss at low SNR. This behavior is inherent in a loop that removes
>modulation -- it's not the fault of either the Costas or squaring loop
>design.
>
>Phil
>
>

Phil have you ever tried to see what the minimum SNR is for a computer simulation of a PLL ? I see in Gardner that he says that at low SNR's the PLL behaves differently due to non-linearities in the practical hardware. But in you computer simulations one can perhaps implement an ideal phase detector etc....

Gardners says (I hope I understand correctly) that a typical PLL will hold in down to 0dB but acquire only at about 3 to 6dB input CW SNR.

Regards, Danie

22. Re: [HFSIG:1255] Re: PSK Question..

From: zs6awk@global.co.za (Danie Brynard)
Date: Sun, 07 Jul 1996 15:43:04 +0200
To: hfsig@tapr.org
Message-ID: <199607071741.PAA22280@lin01.global.co.za>
Rendered body type: text/plain

Tom I found some nice explanations in Gardner. Busy reading..

73 danie

>> 2) What is the minimum signal to noise required for the following BPSK
>> demodulator techniques:
>>
>> - squaring loop PLL type
>> - PSK Costas Loop
>> - or perhaps what is the minimum SNR for a PLL to maintain phase lock ?
>
>   This is a difficult question to answer analytically, and I
>think empirical results are more trustworthy. I think that the answer
>is implementation dependent. No good references to give you right now.
>
>
>   - Tom
>
>
>Tom McDermott
>mcdermot@aud.alcatel.com
>
>

23. Re: [HFSIG:1268] vco block for simulink..

From: chbrain@dircon.co.uk (Charles Brain)
Date: Sun, 07 Jul 1996 18:05:26 +0100
To: hfsig@tapr.org
Message-ID: <199607071714.AA20578@felix.dircon.co.uk>
Rendered body type: text/plain

>Does somebody know if simulink has a vco block and if not where can one get
>it ? I am considering to try a modem simulation on simulink but from what I
>hear there is no vco. (in the older versions in any way)
>
>regards, danie
>
>
I am afraid I can't help you there Danie but PC-SIM has one and I
managed to prove to myself that Costas Loops work !

73 Charles.

24. Q^2PSK

From: forrerj@peak.org (Johan Forrer)
Date: Sun, 07 Jul 1996 10:30:00 -0800
To: HFSIG@tapr.org
Message-ID: <199607071726.KAA17332@PEAK.ORG>
Rendered body type: text/plain

Hi,

Has anyone heard of Q^2PSK (quadrature-quadrature phase shift keying). This
is a four dimensional constellation, i.e., both I and Q being QPSK
modulated. It may be visualized as a hypercube with the four constellation
points at it's corners. The waveform has a constant amplitude.

I find very little about it in books, but have two papers about how it works
from the British IEE Journal.

Any further information will be appreciated.

--Johan

25. Re: [HFSIG:1271] Re: vco block for simulink..

From: Jim Sanford <JSANFORD@INFI.NET>
Date: Sun, 07 Jul 1996 14:31:25 -0400
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19960707183125.23efa698@infi.net>
Rendered body type: text/plain

I'm about to order pc-sim . .. can you share your file as a "learning" tool?

Thanks & 73, Jim wb4gcs@amsat.org

At 12:17 PM 7/7/96 -0500, you wrote:
>>Does somebody know if simulink has a vco block and if not where can one get
>>it ? I am considering to try a modem simulation on simulink but from what I
>>hear there is no vco. (in the older versions in any way)
>>
>>regards, danie
>>
>>
>I am afraid I can't help you there Danie but PC-SIM has one and I
>managed to prove to myself that Costas Loops work !
>
>73 Charles.
>
>
>

26. Re:[HFSIG:1267] Re: UNKN422.WAV

From: Kok Chen <kchen@gala.apple.com>
Date: Sun, 7 Jul 1996 11:41:23 -0700
To: hfsig@tapr.org
Message-ID: <199607071841.LAA20627@gala.apple.com>
Rendered body type: text/plain

Johan (hi, Johan) wrote:

> One other thing that you may give some thought; If the mystery CW
> is machine generated, it should have fairly accurate timing on the
> rising and falling edges. That then brings you to try sequency
> theory, i.e, doing FHT's (Fast Hademard Transforms) that may help
> recovering timing.

Sequency filters (for example, described in H. F. Harmuth,
"Nonsinusoidal Waves for Radar and Radio Communication," Academic
Press, 1981) might work for decoding the _demodulated_ CW, but may
not demodulate the mystery signal well, if the mystery signal was
not using a Walsh function carrier (the FCC would frown on that,
but Harmuth, and creatures who live in a Walsh space would love
you).

Using Hadamard transforms on the original UNKN422.WAV file may
cause all sorts of spurs. Just as square waves create all sorts of
spurs in a Fourier spectrum, a CW (presumably, sinusoidal) carrier
would create a whole series of "Walsh-spurs" in a Hadamard spectrum.

So, one has to still optimally "detect" or demodulate the recorded
signal first, after which, _decoding_ using sequency filters may be
an interesting technique.

If one of us succeeds in decoding the UNKN422.WAV signal, it might
help put the wetware-firmware debate on wreck.radio.amateur.policy
to rest, HI.

73

Kok Chen, AA6TY   kchen@apple.com
Apple Computer, Inc.

P.S. the Harmuth book has some interesting paragraphs on the Barker
sequence that was earlier discussed on HFSIG. In fact, it mentions
the "Complementary Codes" (sum of autocorrelation of the two
"complementary" sequences have no sidelobes). Nice advantage of
complementary codes is that it is not restricted to a maximum (known)
length of 13. Ref: Golay, M. J., "Complementary Series", IRE Trans.
Info. Theory, Vol 7, pp 82-87, 1961. GL es 73.

27. Re: [HFSIG:1272] Q^2PSK

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Mon, 8 Jul 1996 07:56:03 -0400
To: hfsig@tapr.org
Message-ID: <1E0F8BB0.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

Johan,

     This sounds interesting! Could you please let me know what the
     articles are?

73s de
Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1272] Q^2PSK
Author: hfsig@tapr.org at BALT.SMTP
Date: 7/7/96 12:38 PM

Hi,

Has anyone heard of Q^2PSK (quadrature-quadrature phase shift keying). This is a four dimensional constellation, i.e., both I and Q being QPSK modulated. It may be visualized as a hypercube with the four constellation points at it's corners. The waveform has a constant amplitude.

I find very little about it in books, but have two papers about how it works from the British IEE Journal.

Any further information will be appreciated.

--Johan

28. QPSK Channel Spacing

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 08 Jul 1996 12:45:53 -0800
To: HFSIG@tapr.org
Message-ID: <199607081942.MAA27283@PEAK.ORG>
Rendered body type: text/plain

Hi,

Thought I would share a bit feedback after my previous posting regarding why a system of orthogonally-spaced non-coherent QPSK channels does not seem to work very well.

Just by co-incidence, I found the topic discussed in one of my favorite books; Please see Benedetto et. al: Digital Transmission Theory (ISBN 0132143135).

In essence, there is co-channel interference that degrades an orthoganally-spaced system. On page 291 he gives an analysis for channel spacing needed for a BER of 10-6 at a specified Eb/N0.
        For example, receiver bandwidth is optimal at 1.1*T which corresponds to a needed Eb/N0 of 13 dB.
        On an equal basis, he shows Eb/No required for BER of 10-6 for different channel spacings. It is evident that for orthoganal channel spacing (1.0*T), Eb/N0 needs to be in excess of 20 dB. He shows a significant improvement when transmitter bandwidth is limited to 2.4*T, in which case a channel spacing of about 2.5*T needs an Eb/N0 of 12dB for BER of 10-6. These spacings are further dependant whether the channels are keyed in synchronous or asynchronous fashion.

(Note: T=symbol rate, i.e., 75 baud which corresponds to 150 bps for QPSK).

Just for sake of comparison:

Pactor II uses two parallel PSK waveforms each at 100 symbols/sec. According to above observations, this nicely fits into 2*(2.4*T)=480 Hz but would require an Eb/N0=12dB for a BER of 10-6.

The scheme that I propose use channel spacing of (2.0*T) that fits four channels in 4*(2.0*75)=600 Hz with similar performance.

--Johan

29. Re: [HFSIG:1252] Re: PC-DSP + PC-SIM

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Mon, 8 Jul 1996 22:54:07 -0700 (PDT)
To: chbrain@dircon.co.uk (Charles Brain)
Cc: hfsig@tapr.org
Message-ID: <199607090554.WAA14979@unix.ka9q.ampr.org>
Rendered body type: text/plain

>Does anyone know where to get a hold of GNUPLOT, I have searched
>the COMPUSERVE archives and all I can find is the manual !

Like all GNU software, it's available on the Internet. Here's one place:

ftp://prep.ai.mit.edu/pub/gnu/gnuplot-3.5.tar.gz

--Phil

30. Re: [HFSIG:1259] Re: PSK Question..

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Tue, 9 Jul 1996 00:24:11 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607090724.AAA15038@unix.ka9q.ampr.org>
Rendered body type: text/plain

Danie,

PLLs, being inherently nonlinear systems, are *extremely* hard to analyze analytically. What analyses do exist generally make linearity assumptions such as S/N ratios large enough to keep the peak phase jitter small. Computer modeling is heavily used to simulate nonlinear performace such as lock-in behavior, and of course for the results to be useful you have to have initial conditions comparable to your own.

If you've found Gardner, then you have *the* standard reference on PLLs. I believe Dr. Andrew Viterbi also published quite a bit of research into PLLs back in the 1960s. If you can find his mid 1960s book on Coherent Digital Communication you might find some further insights.

My previous message mentioned "squaring loss". The math to derive it quantitatively is quite hairy, but it can be qualitatively understood in several ways. One way is to consider that multiplying a noisy signal by itself produces not only the desired signal times signal term but also unwanted terms consisting of noise times signal and noise times noise -- and these have spectral components that will go through the bandpass filter. At high SNR, the noise is small compared to the signal so the signal times signal term will dominate those involving noise, so the squaring loss is small. At low SNR this is not true.

In a "decision directed" Costas loop, you can see squaring loss manifest itself as errors in the decisions. That is, when a constellation point crosses a decision threshold in the demodulator, the loop incorrectly guesses what the original signal was, and thereby comes up with an incorrect estimate of the error signal.

Squaring loss is a serious problem in Costas loop demodulators operating at very low SNRs, such as in conjunction with a convolutional decoder and soft decisions. That's because a very large percentage of the hard-decoded symbols are in error, even though the decoder can easily handle the equivalent soft-decoded symbols.

Phil

31. Re: [HFSIG:1272] Q^2PSK

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Tue, 9 Jul 1996 00:26:11 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607090726.AAA15042@unix.ka9q.ampr.org>
Rendered body type: text/plain

>Has anyone heard of Q^2PSK (quadrature-quadrature phase shift keying). This
>is a four dimensional constellation, i.e., both I and Q being QPSK

Say what?

How do all the dimensions stay orthogonal? It's not possible to go beyond two dimensions on a single carrier and remain orthogonal.

You could add another carrier and QPSK modulate that, but then that would just be OFDM-QPSK.

Phil

32. Re: [HFSIG:1276] QPSK Channel Spacing

From: dona@tridelta.com (Don Adams)
Date: Tue, 9 Jul 1996 08:24:19 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199607091224.IAA05096@sparcy.tridelta.com>

Rendered body type: text/plain

>
>(Note: T=symbol rate, i.e., 75 baud which corresponds to 150 bps for QPSK).
>
>Just for sake of comparison:
>
>Pactor II uses two parallel PSK waveforms each at 100 symbols/sec. According
>to above observations, this nicely fits into 2*(2.4*T)=480 Hz but would
>require an Eb/N0=12dB for a BER of 10-6.

Is this 400 bps total with 2 subcarriers?

>The scheme that I propose use channel spacing of (2.0*T) that fits four
>channels in 4*(2.0*75)=600 Hz with similar performance.

Is this 600 bps total with 4 subcarriers?

33. Re: [HFSIG:1280] Re: QPSK Channel Spacing

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 09 Jul 1996 08:33:21 -0800
To: hfsig@tapr.org
Message-ID: <199607091529.IAA21833@PEAK.ORG>
Rendered body type: text/plain

Don,

>>
>>(Note: T=symbol rate, i.e., 75 baud which corresponds to 150 bps for QPSK).
>>
>>Just for sake of comparison:
>>
>>Pactor II uses two parallel PSK waveforms each at 100 symbols/sec. According
>>to above observations, this nicely fits into 2*(2.4*T)=480 Hz but would
>>require an Eb/N0=12dB for a BER of 10-6.
>
>Is this 400 bps total with 2 subcarriers?
>
>>The scheme that I propose use channel spacing of (2.0*T) that fits four
>>channels in 4*(2.0*75)=600 Hz with similar performance.
>
>Is this 600 bps total with 4 subcarriers?
>
>
>

Depends on what type of PSK:

BPSK sends 1 bit/symbol, or 200 bps with two subcarriers at 100 baud each,
QPSK sends 2 bits/symbol, or 400 bps with two subcarriers at 100 baud each,
8-QPSK sends 3 bits/symbol, or 600 bps with two subcarriers at 100 baud each,
16-QPSK sends 4 bits/symbol, or 800 bps with two subcarriers at 100 baud each.

Both systems handle a variety of these depending on conditions. These bps
rates of course does not take ECC or compression into account. ECC uses
convolutional coding
with varying rates; at good conditions punctured convolutional code rate 7/8
is used that falls back to 1/2 when needed. Pactor II uses a form of
Markov-Huffman data compression, meaning its on-the-fly coding/decoding
table construction similar to LZW.

My proposed system uses 75 baud symbols and it has a "robust" feature where
speed is traded for robustness, i.e., at 8-DPQSK for example, each bit is
repeated in 12 different slots and diversity combining is used. The
effective bps rate is then 75 bps.

It get quite involved when one need to see all the operating modes - there
are several controlling factors that come into play. You may wonder how does
the two ends of the conversation know what mode it's in? That is resolved by
analysing the packet header where it carries the type of modulation,
diversity factor, code rate etc...

--Johan

34. Re: [HFSIG:1278] Re: PSK Question..

From: Kok Chen <kchen@apple.com>
Date: Tue, 09 Jul 1996 09:57:04 -0700
To: hfsig@tapr.org
Message-ID: <31E28F60.16E5@apple.com>

Rendered body type: text/plain

Phil wrote:

> PLLs, being inherently nonlinear systems, are *extremely* hard to
> analyze analytically. What analyses do exist generally make
> linearity assumptions such as S/N ratios large enough to keep the
> peak phase jitter small.

For a nonlinear analysis of PLLs, try the library under author name
Leon O. Chua. I don't have the book with me here, but if someone
is interested, I'll dig it out. Not too old, published within 2-5
years, if memory serves. Title has the word "Chaotic" or something
to do with Chaos theory in it somewhere. Forgot the second author's
name too, sorry (They say memory is the _second_ thing to go!).
Yep, a PLL works like a chaotic system, bifurcations and all!

73

Kok Chen, AA6TY   kchen@apple.com
Apple Computer, Inc.

35. Re: [HFSIG:1282] Re: PSK Question..

From: zs6awk@global.co.za (Danie Brynard)
Date: Tue, 09 Jul 1996 22:32:40 +0200
To: hfsig@tapr.org
Message-ID: <199607100030.WAA24955@lin01.global.co.za>
Rendered body type: text/plain

Hi Kok. Can you give us the reference please ?

danie

>Phil wrote:
>
>> PLLs, being inherently nonlinear systems, are *extremely* hard to
>> analyze analytically. What analyses do exist generally make
>> linearity assumptions such as S/N ratios large enough to keep the
>> peak phase jitter small.
>
>For a nonlinear analysis of PLLs, try the library under author name
>Leon O. Chua. I don't have the book with me here, but if someone
>is interested, I'll dig it out. Not too old, published within 2-5
>years, if memory serves. Title has the word "Chaotic" or something
>to do with Chaos theory in it somewhere. Forgot the second author's
>name too, sorry (They say memory is the _second_ thing to go!).
>Yep, a PLL works like a chaotic system, bifurcations and all!
>
>73
>
>Kok Chen, AA6TY   kchen@apple.com
>Apple Computer, Inc.
>
>

36. Re: [HFSIG:1283] Re: PSK Question..

From: Kok Chen <kchen@apple.com>
Date: Tue, 09 Jul 1996 16:42:32 -0700
To: hfsig@tapr.org
Message-ID: <31E2EE68.4082@apple.com>
Rendered body type: text/plain

Danie, ZS6AWK, wrote:

> Hi Kok. Can you give us the reference please ?

OK, I had to track the book down, a colleague had borrowed it:

Thomas S. Parker, L. O. Chua, "Practical Numerical Algorithms
for Chaotic Systems," Springer-Verlag, N. Y., 1989.
ISBN 0-387-96689-7 (also, apparently ISBN 3-540-96689-7)

The book speaks more genericly of Chaotic equations, and not
specifically of PLLs (only a couple of paragraphs on that).
However, it references the following paper, which may be what
I remembered:

T. Endo, L. O. Chua, "Chaos from phase-locked loops," IEEE Trans.

Circuits and Systems, 987-1003, August 1988.

73

Kok Chen, AA6TY kchen@apple.com
Apple Computer, Inc.

37. RE: [HFSIG:1283] Re: PSK Question..

From: Rick Booth <rick@itron-ca.com>
Date: Tue, 9 Jul 96 17:26:26 PDT
To: hfsig@tapr.org
Message-ID: <Chameleon.836958891.rick@rickb.itron-ca.com>
Rendered body type: text/plain

There is a large pool of PLL work done by W.C Lindsey and his associates/students that is contained in the following ponderous text:

Synchronization Systems in Communication and Control, W.C. Lindsey, Prentice Hall, ISBN 0-13-879957-1

Also, see:
Principles of Coherent Communication, A.J. Viterbi, McGraw-Hill,

and

Phase Locked Loops, Roland Best, McGraw Hill, ISBN 0-07-911386-9

Rick
W6NZK

Rick Booth
'95 900SS SP
E-mail: rick@itron-ca.com
---

38. regarding ...'BPSK sends 1 bit/symbol'

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Wed, 10 Jul 1996 08:58:12 +0200
To: hfsig@tapr.org
Message-ID: <s1e38e2e.020@KIDD.CO.ZA>
Rendered body type: text/plain

regarding ...'BPSK sends 1 bit/symbol'

Johan does 1bit/symbol have the same meaning as 1bit/Hz ? Thus if you have only 1Hz :-) it would be possible but
the bitrate must be slow enough..

danie

39. Re: [HFSIG:1286] regarding ...'BPSK sends 1 bit/symbol'

From: dona@tridelta.com (Don Adams)
Date: Wed, 10 Jul 1996 08:10:27 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199607101210.IAA07451@sparcy.tridelta.com>
Rendered body type: text/plain

>regarding ...'BPSK sends 1 bit/symbol'
>
>Johan does 1bit/symbol have the same meaning as 1bit/Hz ? Thus if you have
only 1Hz :-) it would be possible but
>the bitrate must be slow enough..
>
>danie

Hi Danie I know Johan will have a technically better explanation, but a "real time"
view of phase modulation I use is that frequency is the rate of change of phase.
So the bandwidth depends on how fast U change the phase.
eg If U instantly flip the phase 180 degrees, U get some "sharp corners" or
some high frequency components or wide bandwidth. If U change the
phase a 1/2cycle(180 degrees) per second, then the bandwidth is about 1/2Hz.
This is why they're worried about the envelope of the rate of change of phase,
to control the bandwidth.
over and out...Don wa8qzz

40. Re: [HFSIG:1286] regarding ...'BPSK sends 1 bit/symbol'

From: Johan Forre <forrerj@peak.org>
Date: Wed, 10 Jul 1996 08:58:04 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.91.960710082801.11200A-100000@kira>
Rendered body type: text/plain

Hallo Danie,

Mmmm... No. bits per symbol does not say anything about
the rate and thus occupied bandwidth. One would need to work in the baud
rate to establish that.

--Johan

On Wed, 10 Jul 1996, Danie Brynard wrote:

> regarding ...'BPSK sends 1 bit/symbol'
>
> Johan does 1bit/symbol have the same meaning as 1bit/Hz ? Thus if you have only 1Hz :-) it would be possible but
> the bitrate must be slow enough..
>
> danie
>
>
>

41. VISSIM

From: "Bloom, Jon, KE3Z" <jbloom@arrl.org>
Date: Wed, 10 Jul 1996 16:38:00 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <m0ue5As-000fNDC@mgate.arrl.org>
Rendered body type: text/plain

For some time, I've been using a Windows program called VisSim. This is a general-purpose system simulation program (for both continuous-time and discrete-time systems), although my exposure to it has been mostly in the context of control systems. VisSim allows you to construct a system as functional blocks, then run a simulation.

Anyway, I wanted to mention this in my QEX editorial because you can download a working demo version of the program via the Web. So I went to the VisSim Web site to check that the demo version was still there and found that not only is it still there, they've added a demo communication system simulation add-on package. I've not had a chance to explore this add-on, but I thought I'd mention it here for those who may be interested.

By the way, the demo version of VisSim is limited only in that it cannot save to disk. However, I discovered a neat trick. Since the entire system description is described in blocks you place in the VisSim window, and since VisSim can copy and paste from the clipboard, you can select the entire system of blocks and copy it to the clipboard, then save it from Clipboard Viewer as a .CLP file. Later, you can load the .CLP file into Clipboard Viewer and then paste it back into the demo VisSim. The only thing this doesn't save, far as I can tell, is the simulation set-up data, which you can re-enter in a few seconds. (I haven't tried this technique with the comm system add-on.)

I'd be interested to hear of the results if anyone tries out this add-on.

Oh yeah: it's at http://www.vissim.com/

 -- Jon, KE3Z

42. Re: [HFSIG:1264] FECC Chipsets (and an apology)

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 10 Jul 1996 18:05:28 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607110105.SAA25964@servo.qualcomm.com>
Rendered body type: text/plain

>What I am wondering, with CD's so popular, is it possible to adapt
>CD-ROM chipsets to radio? I searched the web and the newsgroups, and

Probably not. The CD "channel" is quite different from radio, so the

design considerations for coding are different. An even bigger problem is
the extremely high level of integration found in modern CD players,
which makes it hard to get at just the pieces you want.

On the other hand, general purpose CPUs are now getting so fast that
you can do these functions in software at very respectable speeds.

Phil

43. Re: [HFSIG:1276] QPSK Channel Spacing

From: Barry McLarnon VE3JF <bm@lynx.ve3jf.ampr.org>
Date: Thu, 11 Jul 1996 03:07:58 +0000 (GMT)
To: hfsig@tapr.org
Message-ID: <199607110307.DAA17926@lynx.ve3jf.ampr.org>
Rendered body type: text/plain

> Thought I would share a bit feedback after my previous posting regarding why
> a system of orthogonally-spaced non-coherent QPSK channels does not seem to
> work very well.
>
> Just by co-incidence, I found the topic discussed in one of my favorite
> books; Please see Benedetto et. al: Digital Transmission Theory (ISBN
> 0132143135).
>
> In essence, there is co-channel interference that degrades an
> orthoganally-spaced system. On page 291 he gives an analysis for channel
> spacing needed for a BER of 10-6 at a specified Eb/N0.
>   For example, receiver bandwidth is optimal at 1.1*T which
> corresponds to a needed Eb/N0 of 13 dB.
>   On an equal basis, he shows Eb/No required for BER of 10-6 for
> different channel spacings. It is evident that for orthoganal channel
> spacing (1.0*T), Eb/N0 needs to be in excess of 20 dB. He shows a
> significant improvement when transmitter bandwidth is limited to 2.4*T, in
> which case a channel spacing of about 2.5*T needs an Eb/N0 of 12dB for BER
> of 10-6. These spacings are further dependant whether the channels are keyed
> in synchronous or asynchronous fashion.
>
> (Note: T=symbol rate, i.e., 75 baud which corresponds to 150 bps for QPSK).
>
> Just for sake of comparison:
>
> Pactor II uses two parallel PSK waveforms each at 100 symbols/sec. According
> to above observations, this nicely fits into 2*(2.4*T)=480 Hz but would
> require an Eb/N0=12dB for a BER of 10-6.
>
> The scheme that I propose use channel spacing of (2.0*T) that fits four
> channels in 4*(2.0*75)=600 Hz with similar performance.

Interesting... I'll have to try and find that book. I must admit that I
don't understand this "co-channel interference" mechanism that you refer
to... if the signals are truly orthogonal (which they are for
noncoherent demodulation at 1/T channel spacing, if rectangular pulse
shaping is used), where does the interference come from? There are a
number of practical OFDM systems which use the minimum 1/T spacing, and
I haven't come across any reference to performance penalties incurred by
using this spacing... until now, that is. :-) Here are a few references
which may be of interest:

S.B. Weinstein and P.M. Ebert, "Data transmission by frequency division
multiplexing using the discrete Fourier transform", IEEE Trans. Comm.
Tech., vol. COM-19, pp. 628-634, Oct. 1971.

L.J. Cimini, "Analysis and simulation of a digital mobile channel using
orthogonal frequency division multiplexing", IEEE Trans. Commun., vol.
COM-33, no. 7, pp. 665-675, July 1985.

I'll look into this more tomorrow if I get some time...

Barry

--
  Barry McLarnon VE3JF/VA3TCP | Internet: bm@hydra.carleton.ca
  Ottawa Amateur Radio Club | AMPRnet: bm@lynx.ve3jf.ampr.org
  Packet Working Group | Web: http://hydra.carleton.ca

44. Re: [HFSIG:1269] Re: PSK Question..

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 10 Jul 1996 20:38:29 -0700 (PDT)

To: hfsig@tapr.org
Message-ID: <199607110338.UAA27140@servo.qualcomm.com>
Rendered body type: text/plain

>Phil have you ever tried to see what the minimum SNR is for a computer
>simulation of a PLL ? I see in Gardner that he says that at low SNR's the
>PLL behaves differently due to non-linearities in the practical hardware.
>But in you computer simulations one can perhaps implement an ideal phase
>detector etc....

The idealness of the phase detector has nothing to do with it. A PLL is inherently
a nonlinear system at low SNR, and even a perfect software emulation will
slip cycles at low SNRs. Just as a perfect PSK demod will occasionally produce
errors at low SNR.

>Gardners says (I hope I understand correctly) that a typical PLL will hold
>in down to 0dB but acquire only at about 3 to 6dB input CW SNR.

PLL acquisition is another difficult-to-analyze subject. One thing
that fast CPUs make possible is the notion of a whole bunch of virtual
PLLs search the whole state space (frequency and phase) in "parallel",
looking for the one that most closely matches. This way you don't have
to open up your PLL loop bandwidths to speed up acquisition, which inherently
degrades performance.

Phil

45. Johan:

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Thu, 11 Jul 1996 08:19:49 +0200
To: hfsig@tapr.org
Message-ID: <s1e4baee.009@KIDD.CO.ZA>
Rendered body type: text/plain

Johan:
Mmmm... No. bits per symbol does not say anything about the rate and thus occupied bandwidth. One would
 need
to work in the baud rate to establish that.

Ok Yes Johan, in Manchester II coding there is 1 bit per two symbols.

danie

46. Re: [HFSIG:1290] Re: FECC Chipsets

From: jmorriso@bogomips.com (John Paul Morrison)
Date: Wed, 10 Jul 1996 23:30:34 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <m0ueFGk-000TvfC@bogomips.com>
Rendered body type: text/plain

>
> >What I am wondering, with CD's so popular, is it possible to adapt
> >CD-ROM chipsets to radio? I searched the web and the newsgroups, and
>
> Probably not. The CD "channel" is quite different from radio, so the
> design considerations for coding are different. An even bigger problem is
> the extremely high level of integration found in modern CD players,
> which makes it hard to get at just the pieces you want.
>
> On the other hand, general purpose CPUs are now getting so fast that
> you can do these functions in software at very respectable speeds.

Will the Intel MMX architecture help even more?

>
> Phil
>
>

BogoMIPS Research Labs -- bogosity research & simulation -- VE7JPM --
  jmorriso@bogomips.com ve7jpm@ve7jpm.ampr.org jmorriso@ve7ubc.ampr.org
---

47. Re: [HFSIG:1291] Re: QPSK Channel Spacing

From: Frode Weierud <frode@dxcern.cern.ch>
Date: Thu, 11 Jul 1996 09:17:44 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <Pine.ULT.3.91.960711085949.26087A-100000@dxcern.cern.ch>

Rendered body type: text/plain

On Wed, 10 Jul 1996, Barry McLarnon VE3JF wrote:

> Interesting... I'll have to try and find that book. I must admit that I
> don't understand this "co-channel interference" mechanism that you refer
> to... if the signals are truly orthogonal (which they are for
> noncoherent demodulation at 1/T channel spacing, if rectangular pulse
> shaping is used), where does the interference come from? There are a
> number of practical OFDM systems which use the minimum 1/T spacing, and
> I haven't come across any reference to performance penalties incurred by
> using this spacing... until now, that is. :-) Here are a few references
> which may be of interest:

I too don't quite see the problem if the signals are truly orthogonal,
but the problem is of course that in a medium like HF it is probably
difficult to maintain the true orthogonality due to multipath effects and
Doppler shift. This is of course the reason why most if not all
multi-tone modems on HF use guard bands, to avoid sampling in the head
and tail of each symbol.

Thanks for the references. I have heard about the one from Weinstein and
Ebert and I will have a look for them in the library.

Here is yet another reference to modern OFDM modulation.

Yiyan Wu and William Y. Zou, "Orthogonal Frequency Division Multiplexing:
A Multi-Carrier Modulation Scheme", IEEE Transactions on Consumer
Electronics, Vol. 41, No. 3, Aug. 1995, pp. 392-399.

They also talk about the problem posed by multipath effects and the
resulting ISI. Guard bands is the proposed solution.

73, Frode, F/LA2RL

Frode Weierud   Phone : 41 22 7674794
CERN, SL, CH-1211 Geneva 23,   Fax   : 41 22 7679185
Switzerland   E-mail : Frode.Weierud@cern.ch

48. Re: [HFSIG:1289] VISSIM

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Thu, 11 Jul 1996 08:01:39 -0400
To: hfsig@tapr.org
Message-ID: <1E4EEEB0.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

What is VisSim for? DSP simulation? Generic simulator?

73s de
Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1289] VISSIM
Author: hfsig@tapr.org at BALT.SMTP
Date: 7/10/96 3:02 PM

For some time, I've been using a Windows program called VisSim. This is a
general-purpose system simulation program (for both continuous-time and
discrete-time systems), although my exposure to it has been mostly in the
context of control systems. VisSim allows you to construct a system as
functional blocks, then run a simulation.

Anyway, I wanted to mention this in my QEX editorial because you can
download a working demo version of the program via the Web. So I went to
the VisSim Web site to check that the demo version was still there and
found that not only is it still there, they've added a demo communication
system simulation add-on package. I've not had a chance to explore this
add-on, but I thought I'd mention it here for those who may be
interested.

By the way, the demo version of VisSim is limited only in that it cannot
save to disk. However, I discovered a neat trick. Since the entire system
description is described in blocks you place in the VisSim window, and
since VisSim can copy and paste from the clipboard, you can select the
entire system of blocks and copy it to the clipboard, then save it from
Clipboard Viewer as a .CLP file. Later, you can load the .CLP file into
Clipboard Viewer and then paste it back into the demo VisSim. The only

thing this doesn't save, far as I can tell, is the simulation set-up data, which you can re-enter in a few seconds. (I haven't tried this technique with the comm system add-on.)

I'd be interested to hear of the results if anyone tries out this add-on.

Oh yeah: it's at http://www.vissim.com/

-- Jon, KE3Z

49. Digital voice

From: "Bergzen Hakan, KARL" <hakan.bergzen@enator.se>
Date: Thu, 11 Jul 1996 11:10:00 +0100
To: hfsig <hfsig@tapr.org>
Message-ID: <31EFF57D@noak>
Rendered body type: text/plain

Hello all,
I just wondered whether anybody out there is experimenting with digital voice by using these moderate-to-high speed modems being discussed/develped in this newsgroup.

Since the LPC10e algorithm requires 2400 bps, several of the modems being developed ought to be usable. Especially since it is somewhat error tolerant (therefore the possibility of using parallel tone modems with their assymptotically constant BER at high SNR's), thus no need for extensive FEC. I know of one company that has developed a reduced LPC10e vector set which only requires 600 bps. I haven't heard how it sounds yet though. Has anybody implemented the LPC10e in a standard low-cost DSP (e.g.TMS320xx or DSP56001), or even on a PC?

Another route would be to use the digital voice modules now being embedded in the web reader programs (or as extensions thereof). One of these, from VoxWare, claims to use only 2400 bps. Would it be possible to hook it up to an HF modem, and if so, has anybody tried?

Regards,
Hakan (hakan.bergzen@enator.se)

50. (no subject)

From: Reiner Goeppert <goeppert@ruhp3.physik.uni-freiburg.de>
Date: Thu, 11 Jul 96 15:07:01 METDST
To: hfsig@tapr.org
Message-ID: <199607111306.IAA27193@tapr.org>
Rendered body type: text/plain

--

e-mail: goeppert@ruhp8.physik.uni-freiburg.de
www: http://ruhp8.physik.uni-freiburg.de/goeppert/goeppert_home.html
packet-radio: dl1ggq@db0frb.#bw.deu.eu

51. RE: [HFSIG:1296] Re: VISSIM

From: "Bloom, Jon, KE3Z" <jbloom@arrl.org>
Date: Thu, 11 Jul 1996 12:08:00 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <m0ueNkm-000f49C@mgate.arrl.org>
Rendered body type: text/plain

LANIER.R.A- wrote:
> What is VisSim for? DSP simulation? Generic simulator?

Generic would be a good way to say it. I suggest you look at the web paghe (http://www.vissim.com/), which is much more descriptive than I can be.

-- Jon

52. Re: [HFSIG:1295] Re: QPSK Channel Spacing

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 11 Jul 1996 09:27:52 -0800
To: hfsig@tapr.org
Message-ID: <199607111642.JAA04535@PEAK.ORG>
Rendered body type: text/plain

Hi Barry/Frode,

>On Wed, 10 Jul 1996, Barry McLarnon VE3JF wrote:
>
>> Interesting... I'll have to try and find that book. I must admit that I
>> don't understand this "co-channel interference" mechanism that you refer
>> to... if the signals are truly orthogonal (which they are for
>> noncoherent demodulation at 1/T channel spacing, if rectangular pulse
>> shaping is used), where does the interference come from?

I think this is the real issue ... the raised cosine pulse in my case,
extends each symbol that it overlaps over *four* symbols total, however, it
conforms to all the requirements of a nyquist pulse. This, however, I
suspect, will have some consequences to the orthoganality of the signals. In
the reference I cited, there is no mention of OFDM, rather a discussion of
FDM in general and what it takes in terms of TX and RX filtering to achieve
a certain BER.

In my last posts of the tests that I did, I hope I did not leave the
impression that the guarded orthoganal signalling did not work?;In fact it
does work very well if you don't apply any form of channel filtering at all.
But we know what that a signal like that sounds like? I did then follow the
procedures outlined for MIL-STD-188 filtering for the guarded pulses and saw
the resultant degrading of the recovered constellations. So, my conclusion
is that yes, it works, but if you apply filtering performance does take a
hit as the channels are not truly orthogonal any longer. The question then
answered by the reference I cited is how to practically live with what you have.

My 2 cents worth - hope it helps a bit.

--Johan

53. Re: [HFSIG:1294] Re: FECC Chipsets

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 11 Jul 1996 19:41:17 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607120241.TAA09374@servo.qualcomm.com>
Rendered body type: text/plain

>> On the other hand, general purpose CPUs are now getting so fast that
>> you can do these functions in software at very respectable speeds.

>Will the Intel MMX architecture help even more?

Probably not for FEC, though it's possible that somebody might figure
out a clever way to exploit it.

For DSP, yes. This is what MMX is specifically designed to do. But it
will be a while before MMX is widespread enough that you can write
software in the expectation that most people will be able to use it.

For those who don't know, the MMX (multimedia extensions) consist of
additional instructions designed to aid in digital signal processing.
The general approach is to perform basic operations on groups of four
fixed-point samples at a time. This is a form of SIMD -- single
instruction stream, multiple data stream. Intel calls the MMX data
format "packed", because it packs four integers into a single word.

Also, the MMX instructions perform arithmetic in a saturating fashion.
That is, if you add 1 to FFFF in 16-bit arithmetic, you get FFFF
instead of wrapping around to zero. This is important to reduce the
otherwise nasty effects of an occasional overflow without having to
follow each and every operation with a compare to detect and handle
overflows.

In the meantime, the Pentium floating point unit already makes a very
capable DSP engine. Although operations can only be done on one pair
of numbers at a time, the floating point unit is pipelined. While it
takes 3 clock ticks to complete a typical FPU instruction, if you
write your code carefully, you can "stage" a new FP instruction on
every clock tick. In other words, you can execute up to three
instructions in parallel at different stages of execution. All of the
basic arithmetic instructions (add, subtract, multiply) can stage one
instruction per clock, which pretty much qualifies the Pentium as a
real DSP.

The other week I was at the Smithsonian Air & Space museum where they
have a Cray-1 supercomputer on display. The sign said it was capable of
a sustained rate of 138 megaflops (million floating point instructions
per second). Well, a 133 MHz Pentium running carefully written code
can achieve one floating point operation per clock, or 133 megaflops.

That makes it comparable to a Cray-1! Most impressive.

Phil

54. Re: [HFSIG:1295] Re: QPSK Channel Spacing

From: Barry McLarnon VE3JF <bm@lynx.ve3jf.ampr.org>
Date: Fri, 12 Jul 1996 04:11:02 +0000 (GMT)
To: hfsig@tapr.org
Message-ID: <199607120411.EAA19141@lynx.ve3jf.ampr.org>
Rendered body type: text/plain

Frode Weierud writes:
> I too don't quite see the problem if the signals are truly orthogonal,
> but the problem is of course that in a medium like HF it is probably
> difficult to maintain the true orthogonality due to multipath effects and
> Doppler shift. This is of course the reason why most if not all
> multi-tone modems on HF use guard bands, to avoid sampling in the head
> and tail of each symbol.

True, but in the analysis Johan refers to, I don't think impairments
introduced by the channel are considered. The answer probably lies in
non-rectangular shaping of the data, which destroys the n/T
orthogonality. I think there is a tradeoff here... the OFDM approach
makes good sense when there are a fair number of carriers (and a long
symbol length), but perhaps raised-cosine weighting and greater
separation than 1/T are more attractive when there are a small number of
carriers, as in Johan's case.

> Thanks for the references. I have heard about the one from Weinstein and
> Ebert and I will have a look for them in the library.
>
> Here is yet another reference to modern OFDM modulation.
>
> Yiyan Wu and William Y. Zou, "Orthogonal Frequency Division Multiplexing:
> A Multi-Carrier Modulation Scheme", IEEE Transactions on Consumer
> Electronics, Vol. 41, No. 3, Aug. 1995, pp. 392-399.

Good reference! Yiyan Wu works for the same organization as I do. :-)
Here's one more:

E.F. Casas and C. Leung, "OFDM for data communication over mobile radio
FM channels - part 1: analysis and experimental results", IEEE Trans.
Communications, vol. 39, no. 5, pp. 783-793, May 1991.

Barry

--
  Barry McLarnon VE3JF/VA3TCP | Internet: bm@hydra.carleton.ca
  Ottawa Amateur Radio Club | AMPRnet: bm@lynx.ve3jf.ampr.org
  Packet Working Group | Web: http://hydra.carleton.ca

55. Re: [HFSIG:1272] Q^2PSK

From: Rolf Sommerhalder <rs@ife.ee.ethz.ch>
Date: Mon, 15 Jul 1996 16:27:31 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199607151427.QAA07776@gillespie.ee.ethz.ch>
Rendered body type: text/plain

> Has anyone heard of Q^2PSK (quadrature-quadrature phase shift keying).
> I find very little about it in books, but have two papers about how it works
> from the British IEE Journal.

Try (though not available to me):

[1] D. Saha, "Quadrature-quadrature phase shift keying", Ph.D. dissertation,
    Univ. Michigan, Aug. 1986.

[2] ---, "Quadrature-quadrature phase shift keying", U.S. Pat. 4 680 777,
    July 1987.

[3] ---, "Q2PSK - A new spectrally efficient modulation scheme", presented
    at the 21st Ann. Allerton Conf. Commun., Contr., Comput., Univ. Illinois,
    Oct. 1983.

These refs are from:

[4] D. Saha, T. G. Birdsall, "Quadrature-quadrature phase shift keying",
    IEEE Trans. Comm., Vol 37, No. 5, May 1989.

or more recent papers about Q2PSK with more extensive reference lists
(leading to most of the previously published papers on Q2PSK):

[5] I. Korn, L. Wei, " Q2DPSK in the Satellite Mobile Channel with ISI and CSI",
    ICC Geneva, 1993.

but be careful: the next paper [6] states that a claim in [4] is a fallacy !

[6] H. Sari, "A generalization of multidimensional modulation",
    1995 IEEE International Conference on Communications ICC '95: June 18-22,
    1995, Sheraton Seattle, Piscataway, NJ : IEEE Service Center, cop. 1995.

Hope this helps.

73,
Rolf

56. Re: [HFSIG:1303] Re: Q^2PSK

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 15 Jul 1996 19:35:35 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607160235.TAA12604@servo.qualcomm.com>
Rendered body type: text/plain

Here's the abstract to patent 4,680,777:

Methods for modulating and demodulating digital data streams utilize a
quadrature-quadrature phase shift keying data transmission arrangement
to achieve a 100% increase in the bandwidth efficiency over known
systems, such as minimum shift keying. Known arrangements utilize two
dimensional data transmission. However, Q(^2) PSK, in accordance with
the invention, provides four dimensional transmission which doubles
the rate of data transmission for a given bandwidth, at the expense of
approximately 45% increase in the average energy per bit. The input
data stream is demultiplexed to form four demultiplexed data streams
which are formed of demultiplexed data bits. Each such data stream is
combined with a signal having carrier and data pulse-shaping
components. Additionally, the data pulse-shaping components have a
quadrature phase relationship with each other, thereby adding the
additional two dimensions of data transmission capacity.

Note the mention of a per-bit energy increase of 45%. This clearly
implies that there's a cost for adding these two so-called
"dimensions" that's not present when true multidimensional modulation
is used (e.g., M-ary orthogonal FSK). In other words, the 4 so-called
dimensions of Q-QPSK are not truly orthogonal, nor can they be without
additional bandwidth. It also implies that this modulation is not
particularly well suited to the power-limited channel, which in my
opinion ought to include just about all of radio if it weren't for
artificial bandwidth limits.

Even on the bandwidth limited channel Q-QPSK must be compared to other
16-ary modulation schemes such as 16-PSK and 16-ary QAM. Offhand I
don't know what these other two schemes require in terms of extra
Eb/N0 over perfect QPSK, but I wouldn't be surprised if they're all
comparable to or even better than Q-QPSK.

Phil

57. Q QPSK or not Q QPSK ?

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 16 Jul 96 12:05:41 +0100
To: hfsig@tapr.org
Message-ID: <H0000292020ec174@MHS>
Rendered body type: text/plain

> Here's the abstract to patent 4,680,777:
>
> Methods for modulating and demodulating digital data streams utilize a
> quadrature-quadrature phase shift keying data transmission arrangement
> to achieve a 100% increase in the bandwidth efficiency over known
> systems, such as minimum shift keying. Known arrangements utilize two
> dimensional data transmission. However, Q(^2) PSK, in accordance with
> the invention, provides four dimensional transmission which doubles
> the rate of data transmission for a given bandwidth, at the expense of

> approximately 45% increase in the average energy per bit. The input
> data stream is demultiplexed to form four demultiplexed data streams
> which are formed of demultiplexed data bits. Each such data stream is
> combined with a signal having carrier and data pulse-shaping
> components. Additionally, the data pulse-shaping components have a
> quadrature phase relationship with each other, thereby adding the
> additional two dimensions of data transmission capacity.

I think I understand now!

This reminds me of PACTOR II. Except only one frequency is used.

A pactor II symbol consists of two shaped pulses, roughly rased cosine (not sure what the exact shape is). Each pulse is 1 symbol of one of the two QPSK streams. The two pulses are overlayed so that one reaches a peak when the other reaches a trough, minimising interference between the two streams. In pactor II the two streams are also seperated in frequency to further reduce interference. Clover II is similar except four pulses, on four frequencies are used.

Perhaps QQPSK simply uses the same frequency for the two QPSK streams, but with pulse shapes still out of phase. This could be viewed as a form of TDM, or more simply QPSK with symbols wider than the bit period (reducing bandwidth), but with pulse shapes phased such that there is little ISI. But is ISI really reduced? Over simply a square pulse twice as wide, yes, but I think ISI will still be significant, reducing Eb/No performance, hence the need for more Eb.

If this is really what QQPSK is, then it's not so mysterious, and not truely 4 dimentional. But it is an interesting way of applying pulse shaping and deliberate ISI to trade off Eb/No for a bandwidth reduction.

However I have heard of true 4 dimentional modulation in optics. The extra dimentions come from polarisation modulation. The four dimentions are I, Q, clockwise, anticlockwise (independent circular polarisations like + and - frequency, not antipodal). I recall a seminar back at university where Professor Desmond Taylor (a real fan of Veterbi and Ungerboch (sp?) coding) was explaining his 4 dimentional constalation and the trellis coding scheeme he was using to get optimal Veterbi decoding in 4D. At the time most of it went way over my head. I recall that the physical implementation of it involved mechanical devices that stretched various fiber paths to sync to the light beam. Weird stuff.

Cheers,

Rob, G0VTQ

58. Re: [HFSIG:1305] Q QPSK or not Q QPSK ?

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 17 Jul 1996 18:16:49 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607180116.SAA13777@servo.qualcomm.com>
Rendered body type: text/plain

>little ISI. But is ISI really reduced? Over simply a square pulse twice
>as wide, yes, but I think ISI will still be significant, reducing Eb/No
>performance, hence the need for more Eb.

Note that ISI cannot be overcome by increasing Eb/N0 -- it must either be designed out with proper tx filtering, or compensated for at the rx with an equalizer. In some cases, deliberate ISI (e.g., duobinary) can be decoded with the Viterbi algorithm.

>However I have heard of true 4 dimentional modulation in optics. The
>extra dimentions come from polarisation modulation. The four dimentions
>are I, Q, clockwise, anticlockwise (independent circular polarisations
>like + and - frequency, not antipodal). I recall a seminar back at

Yes, this would be a true four-dimensional scheme, and would not require any Eb/N0 penalty.

Don't let the notion of many dimensions warp your brain too much. "Dimensions" in modulation are just a fancy way to refer to setting channels that are independent of each other, i.e., orthgonal.

Independence between multiple dimensions can be acquired in many ways: by the use of orthogonal basis functions in time (e.g., two BPSK signals in phase quadrature on the same frequency producing QPSK, FDM with sufficient carrier spacing to ensure orthogonality, or

nonoverlapping time slots as in Clover or Pactor II), or use of
physical means such as cross polarization.

All of these schemes (except cross polarization, which you can only
use once, i.e., to create two dimensions) exact their toll as you
increase the number of dimensions: they use extra bandwidth. This is
fundamental -- a scheme that pretends to add orthogonal dimensions
without using extra total bandwidth (or reducing the bandwidth per
orthogonal dimension) is NOT truly orthogonal, and MUST use a higher
Eb/N0 to achieve the same bit error rate.

I have my doubts about schemes like Clover and Pactor II which walk
sequentially through a series of carriers (4 for Clover, 2 for Pactor
II). I understand why they do it -- to let the ISI die down before
sending the next symbol on the same frequency channel. It's
essentially a simplified form of frequency chirping, which is itself a
form of frequency hopped spread spectrum, and that's a standard way to
deal with multipath spread. But it would probably be better if they
started by increasing (beyond 2) the number of orthogonal dimensions
in their signal constellation, e.g., by going to M-ary FSK before they
hop. That would cut down the symbol rate and the effects of ISI.

Phil

59. Re: [HFSIG:1301] Re: FECC Chipsets

From: jmorriso@bogomips.com (John Paul Morrison)
Date: Wed, 17 Jul 1996 18:43:31 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <m0ugi7n-000TutC@bogomips.com>
Rendered body type: text/plain

> In the meantime, the Pentium floating point unit already makes a very
> capable DSP engine. Although operations can only be done on one pair
> of numbers at a time, the floating point unit is pipelined. While it
> takes 3 clock ticks to complete a typical FPU instruction, if you
> write your code carefully, you can "stage" a new FP instruction on
> *every* clock tick. In other words, you can execute up to three
> instructions in parallel at different stages of execution. All of the
> basic arithmetic instructions (add, subtract, multiply) can stage one
> instruction per clock, which pretty much qualifies the Pentium as a
> real DSP.

Does the FPU run indepently of the CPU - ie can the CPU do other
things, like schedule processes in a multi-tasking/multi-threaded
OS, or fetch data to feed into the FPU? I'm just wondering whether
"Native" Signal Processing hogs the whole processor, or whether
you can keep other applications running.

Anyone notice this: for some reason P133's "feel" a lot faster
than their clock speeds would indicate. Maybe it's just better
caching/motherboard design.

BogoMIPS Research Labs -- bogosity research & simulation -- VE7JPM --
  jmorriso@bogomips.com ve7jpm@ve7jpm.ampr.org jmorriso@ve7ubc.ampr.org
---

60. Re: [HFSIG:1269] Re: PSK Question..

From: Bob McGwier <n4hy@ccr-p.ida.org>
Date: Thu, 18 Jul 1996 11:14:33 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199607181514.LAA10408@growler.ccr-p.ida.org>
Rendered body type: text/plain

I am several days removed from this discussion as I have been backpacking in
the Sangre de Cristos in New Mexico, but I have something to add at this
late date. When the signal to noise ratio is low, you cannot even come close
to making the "small angle" phase error assumption used to linearize the
phase detectors and loops. You then truly have some serious nonlinearities
and one MUST make use of the observational evidence during these periods to
even come close to decent performance. The maximum likelihood estimator under
white noise conditions (say a weak signal coming from space) is a nonlinear
filtering estimator and smoother as a posteriori estimator of the signal state.
It only took me four years of training for my Ph.D. in mathematics (stochastic
processes and probability theory) to write a thesis on this topic. The
sufficient statistic is a infinite dimensional estimator on a function space
or Wiener path space and you can forget using it in an intelligent fashion.

All is not lost however.

You can make working suboptimal estimators that use some of the nonlinearities
and I suggest that the best one for this group to consider is a version of
the extended Kalman filter. It is the "linearized Phase lock loop" where you
rederive the linearization at each estimate. If this proves insufficient to
the task you can interate the procedure. That is, you derive the extended
Kalman filter estimate in the forward time direction and use the new estimate
of the state to rederive the "gains", innovations update, etc. and interate
until you are satisfied (this can be until it "stops moving" or has been done
a fixed or maximum number of times). The same can be done for the smoother
going "backwards" in time and the two combine to make a practical if suboptimal
a posteriori estimator.   Since most DSP implementations cannot afford to
keep the entire signal in the computer, some practical length must be forced
on the forwards and backwards estimators for your "a posteriori" estimator.

I wrote this hurriedly, and hope that at least get the flavor of what I am
talking about.

Bob

61. Re: [HFSIG:1306] Re: Q QPSK or not Q QPSK ?

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 18 Jul 1996 08:12:53 -0800
To: hfsig@tapr.org
Message-ID: <199607181526.IAA04851@PEAK.ORG>
Rendered body type: text/plain

Phil,

>
>
>I have my doubts about schemes like Clover and Pactor II which walk
>sequentially through a series of carriers (4 for Clover, 2 for Pactor
>II).

I'm not sure Pactor II falls in this category. As far as I know and the way
I'm implementing their protocol, is that its simply two parallel DQPSK
channels. There may be some phase adjustments for controlling crest factor,
but not at all like Clover that uses both frequency and time interleaving
such as described below.

If anyone has seen real Pactor II waveform specifications or know that it's
not straight DQPSK, please correct me.

>I understand why they do it -- to let the ISI die down before
>sending the next symbol on the same frequency channel. It's
>essentially a simplified form of frequency chirping, which is itself a
>form of frequency hopped spread spectrum, and that's a standard way to
>deal with multipath spread. But it would probably be better if they
>started by increasing (beyond 2) the number of orthogonal dimensions
>in their signal constellation, e.g., by going to M-ary FSK before they
>hop. That would cut down the symbol rate and the effects of ISI.

This is quite interesting that you compare it to a form of frequency chirping.
When one take an OFDM ensamble for example, and then adjust phases for
minimal crest factor with some method like for example, Newman sequences, it
produces a time-domain pulse that has striking resemblances to an up-chirp
with a nearly flat amplitude.

--Johan

62. Digital Communications Conference

From: Rick Whiting <rwhiting@winternet.com>
Date: Thu, 18 Jul 1996 20:25:41 -0500 (CDT)
To: ham-digital@ucsd.edu, hfsig@tapr.org, netsig@tapr.org
Message-ID: <199607190125.UAA19842@absolut-zero.winternet.com>
Rendered body type: text/plain

1996 ARRL and TAPR Digital Communications Conference
September 20-22, 1996
Seattle, Washington (minutes from SeaTac airport)

Web: http://www.tapr.org

It's that time again! Time to make your travel plans and put the finishing

touches on your work for the upcoming 15th Annual ARRL and TAPR Digital Communications Conference. This year marks the first year in which the ARRL Digital Communications Conference and TAPR Annual General Meeting have joined into one conference!

The ARRL and TAPR Digital Communications Conference is an international forum for radio amateurs in digital communications, networking, and related technologies to meet, publish their work, and present new ideas and techniques for discussion. Presenters and attendees will have the opportunity to exchange ideas and learn about recent hardware and software advances, theories, experimental results, and practical applications. The Digital Communications Conference is not just for the digital expert, but for digitally-orientated amateurs of all levels of experience.

The 1996 ARRL and TAPR Digital Communications Conference will be held on September 20-22, 1996 in Seattle, Washington. This year's conference location is just minutes away from the SeaTac (Seattle/Tacoma) Airport.

Not only is the Digital Communications Conference technically stimulating, it is a weekend of fun for all who have more than a casual interest in any of the ham digital communications modes. This includes BBS operators, networkers, DX-Cluster Sysops, software writers, modem designers, and digital satellite communications enthusiasts. The ARRL and TAPR Digital Communications Conference is for all levels of digital operators -- a must conference to attend to get active on a national level. Now, more than ever, amateur radio needs this great meeting of the minds, since it is important that we demonstrate a continued need for the frequency allocations we now have by pushing forward and documenting our achievements. The ARRL and TAPR Digital Communications Conference is one of the few ways to record our accomplishments and challenge each other to do more.

A Conference for the Beginner as well

The conference is not just for the digital expert. This year's conference will again provide an entire morning with beginning and intermediate presentations on selected topics in digital communications. Some of the topics will include: APRS, Satellite Communications, TCP/IP, Digital Radio, Spread Spectrum and other introductory topics. Come to the conference and hear these topics presented by the experts! Don't miss this opportunity to listen and talk to others in this area.

Workshops

In addition to the presentation of papers on Friday and Saturday, three workshops will be held during the conference. On Friday, Keith Sproul, WU2Z, will hold a workshop on APRS packet-location software. Keith is the Chair of the TAPR APRS Special Interest Group, developer of the Macintosh and more recent co-developer of the Windows95 version of APRS, and a leader in the area of APRS technology. This is a unique opportunity to gain insight into this fast growing new digital aspect of amateur operations that combines computers, packet radio, and GPS (Global Positioning Satellites). On Sunday, Dewayne Hendricks, WA8DZP , will conduct a workshop focusing on "How to utilize Part 15 wireless Radios for Ham Applications." Dewayne is an expert in the area of commercial wireless systems; his company WarpSpeed Imagineering, focuses on wireless Internet connectivity. This workshop presents an opportunity to learn how Personal Communications Technology (handheld and small business wireless systems) can be used in the amateur service. A second Sunday workshop will focus on Wireless Networking using the WA4DSY 56K RF modem Technology. This workshop will focus on the technology and accessories of creating and maintaining 56K networks using the WA4DSY modem and equipment compatible with it such as routers, digital driver cards, transverters, and repeaters. Use of WA4DSY 56K equipment in the 219-220 band will also be discussed.

1st Annual ARRL and TAPR DCC Student Papers Award

ARRL and TAPR especially welcome papers from full time students to compete for the first annual student papers award. Two $500 travel awards will be given, one in each of the following categories: a) best technical/theory-oriented paper by a student, and b) best educational or community-oriented application paper by a student. The paper should relate directly to a wireless digital communication topic (see guidelines for more information). Papers coauthored by educators or telecommunications

professionals are also eligible for this award, as long as a student is the first author. First year awards have been funded through a grant by The ARRL Foundation, Inc. Deadline for receipt of finished student paper manuscript: June 11, 1996. Please note that this deadline is different than the general conference submission date. For full details and paper guidelines contact TAPR or check http://www.tapr.org.

Call for Papers

Anyone interested in digital communications is invited to submit a paper for publication in the Conference Proceedings. Presentation at the Conference is not required for publication. If you know of someone who is doing great things with digital communications, be sure to personally tell them about this! Papers are due by July 23, 1996, and should be submitted to Maty Weinberg, ARRL, 225 Main Street, Newington, CT 06111 or via the Internet to lweinberg@arrl.org. Information on paper submission guidelines are available on-line (http://www.tapr.org).

Local Co-Hosts

The 1996 ARRL and TAPR Digital Communications Conference is co-hosted by the Puget Sound Amateur Radio TCP/IP Group and Boeing Employees Amateur Radio Society (BEARS).

The Puget Sound Amateur Radio TCP/IP group is an informal group involved in an ongoing project to build and expand an amateur radio digital network in the greater Puget Sound area of the Pacific Northwest US. The Washington Experimenters TCP/IP Network (WETNET) uses TCP/IP as its primary transport protocol and currently has over 250 users. WETNET is linked to other amateur radio TCP/IP networks via the Internet. The Boeing Employees Amateur Radio Society (BEARS) is a general interest amateur radio club for employees of the Boeing Company, headquartered in Seattle, Washington. The BEARS are an active amateur club, supporting radio classes, VHF/UHF repeaters, and digital communications. BEARS has been instrumental in the construction of the Evergreen Intertie, an extensive network of interconnected repeaters in the Pacific Northwest.

What can you expect during the 1996 ARRL and TAPR Digital Communications Conference ?

* A full day of papers and breakouts on Saturday for the beginner to the advanced amateur digital enthusiast.

* Three workshops
  Friday (4pm) - APRS, Conducted by Keith Sproul, WU2Z
  Sunday (8am) - How to utilize Part 15 Radios for Ham Applications, Conducted by Dewayne Hendricks, WA8DZP
  Sunday (noon) - Wireless Networking using the WA4DSY 56K RF modem Technology

* The first annual Student paper session.

* A banquet with Special Guest Speaker Lyle Johnson, WA7GXD
  Lyle was one of the founders of TAPR and instrumental in forming many of the current aspects of Amateur Digital Communications. He is currently very active in building several digital aspects of the upcoming Phase 3D satellite.

* SIGs (Special Interest Groups) on Saturday following the banquet.

* Informal get-togethers throughout the weekend.

* A meeting facility that is perfect for this type of meeting.

* Vendor area and informal engineering discussions/demonstrations.

* An event at which the most important new developments in amateur digital communications are announced.

* Digital 'movers and shakers' from all over the world in attendance.

* Plenty of Washington State hospitality!

Conclusion

If you have attended a Digital Communications Conference in the past, just remember back to how much fun it was discussing the latest developments into the wee hours! If you have never been, then make your plans now to attend and find out how much fun the Digital Communications Conference can be.

There are few activities where the importance of your participation can be so much fun and important! What a great way to share and renew your enthusiasm for digital amateur radio! Getting together with colleagues from all over the world and bringing each other up to date on your latest work. All this, and more, for an unforgettable weekend of ham radio and digital communications. Make your travel and lodging arrangements now. We hope to see you at the ARRL and TAPR Digital Communications Conference on September 20-22!

Full information on the conference and hotel information can be obtained by contacting Tucson Amateur Packet Radio, 8987-309 E. Tanque Verde Road #337, Tucson, AZ 85749-9399. Phone: (817) 383-0000. Fax: (817) 566-2544. Internet: tapr@tapr.org Web: www.tapr.org

Sincerely,
Steve Ford, WB8IMY, ARRL Conference Co-Chair
Keith Justice, KF7TP, TAPR Conference Co-Chair
Steve Stroh, N8GNJ, Local Host Liaison
Greg Jones, WD5IVD, President TAPR

Note: If you need handouts or flyers for meetings, contact TAPR about getting what you need!

Hotel Information

Conference presentations, meetings, and workshops will be held at the Quality Inn Seattle Airport, a complex co-located with the Radisson Hotel Seattle Airport. Rooms rates are $66/single-double and $76/triple. When making reservations with the hotel, be sure to indicate you are attending the ARRL and TAPR DCC conference. It is highly recommended that you book your room prior to arriving - a block of 75 rooms is reserved until September 6th, 1996. After the 75 rooms are booked, rooms will only be available in the Radisson hotel, but will be at a higher price. Be sure to book your rooms early! The hotel provides transportation to and from SeaTac Airport. You should contact the hotel to arrange airport transportation.

Quality Inn Seattle Airport (conference hotel)
17101 Pacific Highway South, Seattle, WA, 98188
(206) 246-7000, Fax (206) 246-1715

Radisson Hotel Seattle Airport (alternate hotel)
17101 Pacific Highway South, Seattle, WA, 98188
(206) 244-6000, Fax (206) 246-6835

Registration

Contact the TAPR office by Phone, Fax, or e-mail (Internet: tapr@tapr.org) to preregister or for additional meeting information. MasterCard and VISA accepted.

- Preregistration (before Sept 1st) $40.00 *

- Late Registration or at door $45.00 *

* - Conference Registration includes:
Conference Proceedings, Sessions, Meetings, and Lunch.

- Saturday Evening Dinner (Limited Space) $19.00 **

** - Dinner, Speaker: Lyle Johnson, WA7GXD, Prize Drawing

Workshops

APRS Workshop
Friday, 4pm - 7pm. Conducted by Keith Sproul, WU2Z

- Registration $15.00

How to utilize Part 15 Radios for Ham Applications Workshop, Sunday, 8:00am
- 11:00am. Dewayne Hendricks, WA8DZP

- Registration $15.00

Wireless Networking using the WA4DSY 56K RF modem Technology Workshop
Sunday, 12noon - 3pm.

- Registration $15.00

Contact TAPR to register for the DCC.
Tucson Amateur Packet Radio, 8987-309 E. Tanque Verde Road #337, Tucson, AZ
85749-9399. Phone: (817) 383-0000. Fax: (817) 566-2544.
Internet: tapr@tapr.org   http://www.tapr.org

  Richard A. (Rick) Whiting   Phone: + 1 612 550 1213
  5780 Rosewood Ln. N.   E-mail: rwhiting@winternet.com
  Plymouth, MN 55442-1411   Packet: W0TN @ WB0GDB.MN.USA.NOAM
   INDEPENDENT PROFESSIONAL CONSULTANT - TELECOMMUNICATIONS & WIRELESS
 ----

63. Are the new amateur datacomms modulations and protocols freely available ? If not why not ? What mus

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Fri, 19 Jul 1996 07:57:19 +0200
To: hfsig@tapr.org
Message-ID: <s1ef410f.002@KIDD.CO.ZA>
Rendered body type: text/plain

Are the new amateur datacomms modulations and protocols freely available ? If not why not ? What must
one do if
you want to write a modem for a new emerging modulation/protocol ?

Danie

64. Re: [HFSIG:1311] Are the new amateur datacomms modulations and protocols

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Fri, 19 Jul 1996 08:50:35 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <Pine.LNX.3.91.960719084458.29452A-100000@bacchus.uet.e-technik.th-darmstadt.de>
Rendered body type: text/plain

On Fri, 19 Jul 1996, Danie Brynard wrote:

> Are the new amateur datacomms modulations and protocols freely available ? If not why not ? What must
one do if
> you want to write a modem for a new emerging modulation/protocol ?
>
> Danie
>

I was talking to the developers of Pactor II the other week and they are
not willing to make the Pactor II specs available freely. I really don't
know any reason for this, but it is their decision. This means - at least
for German amateurs - that it could be illegal to use Pactor II.

73' Alexander

-------------------------------+---------------------------------------
Alexander F. Kurpiers	
Institut fuer Uebertragungs-	Voice: +49-6151-162369
u. Elektroakustik	Fax : +49-6151-165545
Merckstrasse 25	EMail: a.kurpiers@uet.th-darmstadt.de
D-64283 Darmstadt (Germany)	Hamradio: dl8aau@db0zdf.#rpl.deu.eu
-------------------------------+---------------------------------------

65. Alexander wrote..

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Fri, 19 Jul 1996 16:29:01 +0200
To: hfsig@tapr.org
Message-ID: <s1efb8e7.052@KIDD.CO.ZA>
Rendered body type: text/plain

Alexander wrote..
I was talking to the developers of Pactor II the other week and they are not willing to make the Pactor
 II specs
available freely. I really don't know any reason for this, but it is their decision. This means - at
least for German
amateurs - that it could be illegal to use Pactor II.
==========
Ja very interesting. I feel the modulation standard and protocol should be widely published ie in ARRL
Handbook.

danie

66. Re: [HFSIG:1312] Re: Are the new amateur datacomms modulations and protocols

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 19 Jul 1996 08:34:47 -0800
To: hfsig@tapr.org
Message-ID: <199607191548.IAA02480@PEAK.ORG>
Rendered body type: text/plain

Alexander,

That is a thorny question indeed. You are right that the new regulations in
the US also requires that a protocol be "publicly published". I do have a
sense though that Clover and Pactor both have been "grandfathered", i.e.,
by default accepted as legal. You will probably have a hard time obtaining
implementation details beyond the sketcy glossy sales brochures.

If you wish to develop a new scheme, why not discuss it on a forum like
HFSIG - you will get some wonderful ideas and be able to use the knowledge
of outstanding brains in the field. This way it is publicly available and
legal. Further, I do believe that if you have plans to commercialize your
idea, you will gain a lot more by promoting such an open architecture. There
would be plenty of opportunity for innovation and technical advantages to
offer. There is a lot to be said for doing a good job at both hardware and
software that is available at a reason able price. In this field, acceptance
by many users tells when something is successful.

Time permitting at the Digital Communications Conference HFSIG meeting I
will outline the new HF digital communications scheme that I am presently
working on. It's called QUATOR and it will be an open architecture - all
details will be freely available. Make a point to attend the DCC and give
you an opportunity to give some input.

My two cents worth.

--Johan, KC7WW

>On Fri, 19 Jul 1996, Danie Brynard wrote:
>
>> Are the new amateur datacomms modulations and protocols freely available
? If not why not ? What must one do if
>> you want to write a modem for a new emerging modulation/protocol ?
>>
>> Danie
>>
>
>
>I was talking to the developers of Pactor II the other week and they are
>not willing to make the Pactor II specs available freely. I really don't
>know any reason for this, but it is their decision. This means - at least
>for German amateurs - that it could be illegal to use Pactor II.
>
>73' Alexander
>
>
>*-------------------------------+---------------------------------------*
>| Alexander F. Kurpiers | |
>| Institut fuer Uebertragungs- | Voice: +49-6151-162369 |
>| u. Elektroakustik | Fax : +49-6151-165545 |
>| Merckstrasse 25 | EMail: a.kurpiers@uet.th-darmstadt.de |
>| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0zdf.#rpl.deu.eu |
>*-------------------------------+---------------------------------------*

>
>

67. Re: [HFSIG:1313] Alexander wrote..

From: k6sti@n2.net (Brian Beezley)
Date: Fri, 19 Jul 1996 09:50:27 -0700
To: hfsig@tapr.org
Message-ID: <199607191650.JAA22300@ravel.n2.net>
Rendered body type: text/plain

>Alexander wrote..
>I was talking to the developers of Pactor II the other week and they are
not willing to make the Pactor II specs
>available freely. I really don't know any reason for this, but it is their
decision. This means - at least for German
>amateurs - that it could be illegal to use Pactor II.
>==========
>Ja very interesting. I feel the modulation standard and protocol should be
widely published ie in ARRL Handbook.
>
>danie
>

But you didn't expend the resources to develop the algorithm. SCS tells me
they spent a very long time researching and testing the protocol. When you
do that, you can't afford to give the final result away. I'd love to
implement Pactor II myself, but I can understand why licenses aren't being
offered cheap.

73--Brian, K6STI
k6sti@n2.net

68. Re: [HFSIG:1314] Re: Are the new amateur datacomms modulations and protocols

From: Jim Sanford <JSANFORD@INFI.NET>
Date: Fri, 19 Jul 1996 16:12:35 -0400
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19960719201235.2eb739e2@infi.net>
Rendered body type: text/plain

At 10:56 AM 7/19/96 -0500, you wrote:
>Alexander,
>
>That is a thorny question indeed. You are right that the new regulations in
>the US also requires that a protocol be "publicly published". I do have a
>sense though that Clover and Pactor both have been "grandfathered", i.e.,
>by default accepted as legal. You will probably have a hard time obtaining
>implementation details beyond the sketcy glossy sales brochures.

A good reason to support neither Pactor nor Clover by buying the products.

Jim
wb4gcs@amsat.org

69. Re: [HFSIG:1314] Re: Are the new amateur datacomms modulations and protocols

From: Mike Curtis <wd6ehr@kaiwan009.kaiwan.com>
Date: Fri, 19 Jul 1996 14:21:29 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607192121.OAA22484@kaiwan009.kaiwan.com>
Rendered body type: text/plain

> That is a thorny question indeed. You are right that the new regulations in
> the US also requires that a protocol be "publicly published". I do have a
> sense though that Clover and Pactor both have been "grandfathered", i.e.,
> by default accepted as legal. You will probably have a hard time obtaining
> implementation details beyond the sketcy glossy sales brochures.

As I understand it, you could submit an article to QST, QRX, 73, etc., and
this would qualify as "publicly published". I would even hazard the guess
that sending a copy to the FCC and posting something to ALLUS would qualify
as being "publicly published". The ARRL could probably clarify this.

> If you wish to develop a new scheme, why not discuss it on a forum like
> HFSIG

Which might also qualify as being "publicly published".

-- mike

70. Twin Cities

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Fri, 19 Jul 1996 16:27:14 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199607192127.QAA10251@www.kelly-afb.org>
Rendered body type: text/plain

Rick,

Was in the Twin Cities Sun nite thru this morning but didn't have time to
ham as my son was taking us (Susie, WB5WXY and I) out to Manny's, to
Stillwater, other fine resturants and his Buca resturant out in Eden
Prairie. Also took a boot ride from St Paul and of course visited the
"Mall".

Spent too much money so may have to pass on Seattle but I'm still hoping
Mama will let me "charge" a round trip ticket...Hi Hi.

73,

Walt

PS, we're off to Boise, ID in the morning to see my other son and our
granddaughter...we're keeping the airlines in money.

```
=============================================================================
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development.      |
=============================================================================
|                                  |                                  |
|                                  | The greatest dangers to liberty  |
| Walt DuBose - K5YFW              | lurk in insidious encroachment   |
| E-Mail k5yfw@www.kelly-afb.org   | by men of zeal, well-meaning     |
| Business Telephone: (210)925-6081 | but without understanding.      |
|     Home Telephone: (210)696-3196 |                                  |
|                                  |   - Justice Louis D. Brandeis    |
|                                  |                                  |
=============================================================================
```

71. Re: [HFSIG:1317] Re: Are the new amateur datacomms modulations and protocols

From: Greg Jones <wd5ivd@tapr.org>
Date: Fri, 19 Jul 1996 17:02:55 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199607192202.RAA19211@tapr.org>
Rendered body type: text/plain

The ARRL has a publication that discusses the Clover, Pactor and GTOR.
These were presented at the 13th DCC I believe -- which was when the FCC
made the statement about publishing a standard to make it legal

The acceptable method within the US has been to publish it in the Digital
Communication Conference.

Greg

>
>
> > That is a thorny question indeed. You are right that the new regulations in
> > the US also requires that a protocol be "publicly published". I do have a
> > sense though that Clover and Pactor both have been "grandfathered", i.e.,
> > by default accepted as legal. You will probably have a hard time obtaining
> > implementation details beyond the sketcy glossy sales brochures.
>
> As I understand it, you could submit an article to QST, QRX, 73, etc., and
> this would qualify as "publicly published". I would even hazard the guess
> that sending a copy to the FCC and posting something to ALLUS would qualify
> as being "publicly published". The ARRL could probably clarify this.
>
> > If you wish to develop a new scheme, why not discuss it on a forum like
> > HFSIG
>
> Which might also qualify as being "publicly published".
>
>
>

> -- mike
>
>

72. Re: [HFSIG:1318] Twin Cities

From: Charles Hoffman <choffman@pelican.davlin.net>
Date: Fri, 19 Jul 1996 19:35:28 -0500
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19960720003528.0069e128@davlin.net>
Rendered body type: text/plain

Not sure how you will get this Walt! If you do, check out the Southwest Airlines...in a bigtime price war and their tickets are going pretty cheep where ever they service. Glad you are having ooodles of fun...its only moola...enjoy. black walnuts tincture is done FB.

At 16:45 7/19/96 -0500, you wrote:
>Rick,
>
>Was in the Twin Cities Sun nite thru this morning but didn't have time to
>ham as my son was taking us (Susie, WB5WXY and I) out to Manny's, to
>Stillwater, other fine resturants and his Buca resturant out in Eden
>Prairie. Also took a boot ride from St Paul and of course visited the
>"Mall".
>
>Spent too much money so may have to pass on Seattle but I'm still hoping
>Mama will let me "charge" a round trip ticket...Hi Hi.
>
>73,
>
>Walt
>
>PS, we're off to Boise, ID in the morning to see my other son and our
>granddaughter...we're keeping the airlines in money.
>
>===
>| The MicroSoft operating system didn't get as bad as it is overnight,|
>| it has taken over 10 years of careful, calculated development. |
>===
>| | |
>| | The greatest dangers to liberty |
>| Walt DuBose - K5YFW | lurk in insidious encroachment |
>| E-Mail k5yfw@www.kelly-afb.org | by men of zeal, well-meaning |
>| Business Telephone: (210)925-6081 | but without understanding. |
>| Home Telephone: (210)696-3196 | |
>| | - Justice Louis D. Brandeis |
>| | |
>===
>
>
>

73. Re: [HFSIG:1307] Re: FECC Chipsets

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 19 Jul 1996 18:51:37 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607200151.SAA04696@servo.qualcomm.com>
Rendered body type: text/plain

>Does the FPU run indepently of the CPU - ie can the CPU do other
>things, like schedule processes in a multi-tasking/multi-threaded

Read the Intel architecture documents. The FPU is a separate unit in the Pentium, but it executes off the same instruction stream. The Pentium also has two integer pipeline units, although the second can't handle every instruction.

The CPU control logic attempts to keep as many of the units going in parallel as possible as it executes a single instruction stream. So in that sense, no, the FPU doesn't really run independently of the CPU.

Also, each unit can generally work at once on several consecutive instructions in different stages of execution, like an assembly line. This is pipelining. For example, the Pentium floating point unit takes three clocks to finish a multiply (or to do most basic floating point operations other than divide), but because the unit is pipelined it can begin a new multiply instruction before the previous one (or two) have completed.

The CPU's ability to do all this depends heavily on the actual
instruction sequence; that's why the clock counts in the manual vary
so much. Really good compilers know how the Pentium works inside, so
they'll try to emit code that can be readily pipelined. For example,
they'll try to avoid emitting instructions that immediately access the
result of the previous instruction, because that causes a pipeline
"stall" -- the CPU blocks until the needed data is ready before
proceeding, which slows things down. A good compiler will see if it
can do something else instead until the needed data is ready.

Unfortunately, I don't have one of those compilers so for the stuff I
really want to run fast, I'll write hand-optimized assembler designed
to pipeline well. I've done this for a FFT (specifically a radix 4 FFT
butterfly) and for a FIR.

The Pentium Pro has many more performance speedups, including
out-of-order execution of unrelated instructions (meaning the compiler
doesn't have to be as smart to avoid pipeline stalls) and "speculative
execution" of multiple instructions so as to get a head start before
it's known which should actually have been done (e.g., after a
branch). See Intel's web pages for more details.

Phil

---

74. (no subject)

From: (unknown sender)
Date: (date unavailable)
Message-ID: <199607201035.AA14792@felix.dircon.co.uk>
Rendered body type: text/plain

id sma014778; Sat Jul 20 11:35:15 1996
X-Sender: chbrain@popmail.dircon.co.uk
X-Mailer: Windows Eudora Version 1.4.4
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"
Date: Sat, 20 Jul 1996 11:24:31 +0100
To: hfsig@tapr.org
From: chbrain@dircon.co.uk (Charles Brain)
Subject: ECC Book

Has anybody seen this book and if so is it worth adding to the 'library' ?.

Error Coding Cookbook: Practical C/C++ Routines and Recipes for
  Error Detection and Correction
  Disk Included

  C. Britton Rorabaugh

  Hard ISBN 0-07-911720-1 1996
  251p. 71 Illus. 7 3/8 x 9 1/4
  $55.00

                  ** Description **

  At last, the complexities of error coding are made simple If you
  don't happen to be an abstract mathematician, it's all too easy to
  get lost in the maze of error correction coding thats is a crucial
  component of any signal processing or communications system. Now
  this first-of-it-kind, non-theoretical guide clearly explains the
  often complex error coding process. You'll find complete
  step-by-step instructions on how to: Select the most appropriate
  code for your particular application; Design the corresponding
  coders and decoders; Implement many of the most well-known and
  frequently used codes for error correction. Plus, you'll have
  access to ready-to-use C and C++ programs for generating hundreds
  of codes for error detection and correction on the accompanying
  disk. The book's exceptionally clear descriptions of each code are
  reinformed by an array of worked-out examples and a brief review
  of only the necessary math where applicable. There has never been
  a more illuminating or practical treatment of this always
  challenging and occasionally perplexing area.

                  ** Audience **

  Electronics, electrical, communications, and computer engineers
  designing all types of communications and signal processing
  equipment.

** Contents **

Strategic Issues. Mathematical Tools for Coding. Block and Cyclic
Codes. BCH and Reed-Solomon Codes. Encoders and Decoders.
Convolutional Codes. Viterbi Decoding. Sequential Decoding.

75. Re: [HFSIG:1321] Re: FECC Chipsets

From: dibene@VNET.IBM.COM (Alberto di Bene)
Date: Mon, 22 Jul 96 17:22:14 EDT
To: hfsig@tapr.org
Message-ID: <199607221540.KAA08341@tapr.org>
Rendered body type: text/plain

In a previous message, Phil Karn wrote:
>..... snip .....
>
> ..... I've done this for a FFT (specifically a radix 4 FFT
> butterfly) and for a FIR.
>
>..... snip ....
>Phil
>

How about sharing a code snippet with hfsig, showing the butterfly
and the FIR ? The wheel has been reinvented so many times, let's
avoid one more reinvention... :-)

Thanks in any case,

Alberto di Bene, I2PHD dibene@vnet.ibm.com i2phd@amsat.org
---

76. Re: [HFSIG:1312] Re: Are the new amateur datacomms modulations and protocols

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 22 Jul 1996 17:34:15 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607230034.RAA05156@servo.qualcomm.com>
Rendered body type: text/plain

>I was talking to the developers of Pactor II the other week and they are
>not willing to make the Pactor II specs available freely. I really don't
>know any reason for this, but it is their decision. This means - at least
>for German amateurs - that it could be illegal to use Pactor II.

This policy (along with the similar proprietary attitudes of the companies
making Clover and G-TOR) gives us an opportunity to create a new and open
modulation and coding format more in the cooperative spirit of ham radio.
And I believe it could easily outperform these proprietary schemes as well.

Phil

77. Re: [HFSIG:1315] Re: Alexander wrote..

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 22 Jul 1996 18:08:48 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607230108.SAA05265@servo.qualcomm.com>
Rendered body type: text/plain

>But you didn't expend the resources to develop the algorithm. SCS tells me
>they spent a very long time researching and testing the protocol. When you
>do that, you can't afford to give the final result away. I'd love to
>implement Pactor II myself, but I can understand why licenses aren't being
>offered cheap.

Keeping the protocol proprietary isn't necessarily going to pay back
their development costs either. The computer and communications
industry has a long history of proprietary technologies being quickly
displaced by more open, standard systems. Anybody here still use a
Telebit Trailblazer modem in PEP mode?

More enlightened companies are content to have part of something
rather than 100% of nothing.

Phil

78. Re: [HFSIG:1319] Re: Are the new amateur datacomms modulations and protocols

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 22 Jul 1996 18:27:14 -0700 (PDT)

To: hfsig@tapr.org
Message-ID: <199607230127.SAA05321@servo.qualcomm.com>
Rendered body type: text/plain

>The ARRL has a publication that discusses the Clover, Pactor and GTOR.
>These were presented at the 13th DCC I believe -- which was when the FCC
>made the statement about publishing a standard to make it legal

These papers discuss the general approaches used in each scheme, but they
are by no means complete enough to write a compatible modem. So I certainly
wouldn't call them "standards" or "specifications".

Phil

79. Re: [HFSIG:1324] Re: Are the new amateur datacomms modulations and protocols

From: Jim Sanford <JSANFORD@INFI.NET>
Date: Mon, 22 Jul 1996 21:32:30 -0400
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19960723013230.0a07edd8@infi.net>
Rendered body type: text/plain

At 07:51 PM 7/22/96 -0500, you wrote:
>>I was talking to the developers of Pactor II the other week and they are
>>not willing to make the Pactor II specs available freely. I really don't
>>know any reason for this, but it is their decision. This means - at least
>>for German amateurs - that it could be illegal to use Pactor II.
>
>This policy (along with the similar proprietary attitudes of the companies
>making Clover and G-TOR) gives us an opportunity to create a new and open
>modulation and coding format more in the cooperative spirit of ham radio.
>And I believe it could easily outperform these proprietary schemes as well.
>
>Phil
>
           YES, YES, YES!!!!!!!!!!!

Outperform, outsell, obsolete them!

(Especially the guys who use paying hams to test [at a high price] stuff they
want to sell to the military!)

73, Jim
wb4gcs@amsat.org

80. Re: [HFSIG:1326] Re: Are the new amateur datacomms modulations and protocols

From: Greg Jones <wd5ivd@tapr.org>
Date: Mon, 22 Jul 1996 20:48:15 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199607230148.UAA03433@tapr.org>
Rendered body type: text/plain

I agree Phil.

The issue was that these where what the FCC termed as being enough to allow
legal operations under the unspecified digital modes area.

Was not what the FCC agreed to the position that Paul Rinaldo and the Future
Systems Committee work towards ?

But as you point out -- these were not technical specifications.

There is nothing keep this group or amateur radio as a whole from developing
a standard that is more open and works better than current systems.

Greg

>
> >The ARRL has a publication that discusses the Clover, Pactor and GTOR.
> >These were presented at the 13th DCC I believe -- which was when the FCC
> >made the statement about publishing a standard to make it legal
>
> These papers discuss the general approaches used in each scheme, but they
> are by no means complete enough to write a compatible modem. So I certainly
> wouldn't call them "standards" or "specifications".
>
> Phil
>
>

81. Re: [HFSIG:1327] Re: Are the new amateur datacomms modulations

From: Rolf Sommerhalder <rs@ife.ee.ethz.ch>
Date: Tue, 23 Jul 1996 08:17:52 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199607230617.IAA21869@gillespie.ee.ethz.ch>
Rendered body type: text/plain

> >>I was talking to the developers of Pactor II the other week and they are
> >>not willing to make the Pactor II specs available freely. I really don't
> >>know any reason for this, but it is their decision. This means - at least
> >>for German amateurs - that it could be illegal to use Pactor II.
> >
> >This policy (along with the similar proprietary attitudes of the companies
> >making Clover and G-TOR) gives us an opportunity to create a new and open
> >modulation and coding format more in the cooperative spirit of ham radio.
> >And I believe it could easily outperform these proprietary schemes as well.
> >
> >Phil
> >
>   YES, YES, YES!!!!!!!!!!!
>
> Outperform, outsell, obsolete them!
>
> (Especially the guys who use paying hams to test [at a high price] stuff they
> want to sell to the military!)

Unfortunately, it is even worse than selling just to the military: they sell / trying to sell it to the humanitarian aid community (Red Crosses/Crescents, United Nations agencies, non- and government organisations) - and at much higher prices. Though the problems is, that there are currently no decent alternatives really, besides SITOR/AMTOR or maybe GTOR/Pactor-1.

Rolf, HB9CWP

82. Re: [HFSIG:1323] Re: FECC Chipsets

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 23 Jul 1996 02:23:53 -0700 (PDT)
To: dibene@vnet.ibm.com
Cc: hfsig@tapr.org
Message-ID: <199607230923.CAA06324@servo.qualcomm.com>
Rendered body type: text/plain

> How about sharing a code snippet with hfsig, showing the butterfly
> and the FIR ? The wheel has been reinvented so many times, let's
> avoid one more reinvention...   :-)

Okay, I've put the FIR filter on my web page:

http://www.qualcomm.com/people/pkarn/ham.html

I have to clean up and package the FFT code. When I do, I'll add it to the same page.

Phil

83. Re: [HFSIG:1328] Re: Are the new amateur datacomms modulations and protocols

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 23 Jul 1996 02:52:07 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607230952.CAA06370@servo.qualcomm.com>
Rendered body type: text/plain

>The issue was that these where what the FCC termed as being enough to allow
>legal operations under the unspecified digital modes area.

I suspect that all the FCC really cares about is that they can get a box to monitor the emission if they want to. That simply means it has to be commercially available, which they all are.

Unfortunately, that does nothing to promote general appreciation and understanding of what goes on in these newer modulation schemes. And I find that deeply troubling, as it's very much against what I believe to be the amateur radio spirit.

Phil

84. Re: [HFSIG:1309] Re: Q QPSK or not Q QPSK ?

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 23 Jul 1996 03:46:26 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607231046.DAA06525@servo.qualcomm.com>
Rendered body type: text/plain

>I'm not sure Pactor II falls in this category. As far as I know and the way
>I'm implementing their protocol, is that its simply two parallel DQPSK
>channels

You may be right -- I know relatively little about Pactor beyond what I
read in some flyers I picked up at Dayton, and as we know they aren't
publishing full specs.

>This is quite interesting that you compare it to a form of frequency chirping.
>When one take an OFDM ensamble for example, and then adjust phases for
>minimal crest factor with some method like for example, Newman sequences, it
>produces a time-domain pulse that has striking resemblances to an up-chirp
>with a nearly flat amplitude.

So what I can't understand is why people don't just go ahead and build
systems that actually do chirp or spread, instead of building
idiosyncratic approximations of same.

I'm now more convinced than ever that the ideal HF modem would use
M-ary FSK with frequency hopping. And of course the hopping could be
turned off if desired.

Phil

85. HF Modem Bandwidth

From: k6sti@n2.net (Brian Beezley)
Date: Tue, 23 Jul 1996 04:47:20 -0700
To: hfsig@tapr.org
Message-ID: <199607231147.EAA06903@ravel.n2.net>
Rendered body type: text/plain

>BTW, one problem we'll have to face with the scheme I advocate is the
>restrictive bandwidth limit on HF. I strongly advocate a wider signal
>that would permit a considerable reduction in total signal power (10 dB
>or more, according to the figures I've seen for the existing schemes)
>and less QRM overall despite the extra bandwidth.

10 dB is a lot--how much wider are you thinking, Phil? Tell me you don't
want all of 20 meters!

I think to go much beyond 500 Hz in the CW/DATA portion of the bands would
add years to the introduction of a new protocol. The educational effort
required would be enormous. I think you could probably get the FCC to go
along--it's the general ham population that would react very negatively to
any kind of wideband proposal. At the minimum I think wideband signals
would have to be restricted to a tight subband. Still, I'm sure it would
take years to get authorized. I don't have the patience.

You might be able to slip it into the SSB part of the band. I once operated
HF fax with a couple buddies using Western Union Deskfax machines before
this mode was legal on HF. We built little FM modulators, operated near the
SSTV frequencies, and on the air always referred to it as "SSTV." If you
came up with a digital picture-coding scheme that happened to have a great
HF modem built into it and was less than 3 kHz wide, I think you could use
it today on the air without any problem. Just call it Scottie 17 or
something. One day you accidentally send a text file instead of a picture
file and you're on your way. Although this backdoor approach might actually
work, it's a little too sneaky for my taste.

So how wide? And what exactly is the scheme you advocate?

73--Brian, K6STI
k6sti@n2.net

86. RE: [HFSIG:1333] HF Modem Bandwidth

From: Rick Booth <rick@itron-ca.com>
Date: Tue, 23 Jul 96 09:36:00
To: hfsig@tapr.org

Message-ID: <Chameleon.838139912.rick@rickb.itron-ca.com>
Rendered body type: text/plain

Brian,
My guess is the biggest stumbling block to wideband com at HF is the base of transceivers already in place among the ham world...this puts the upper limit at about 2.4 KHz. How many hams will hack a 2 grand plus radio to diddle with wider filters?
Rick
W6NZK

Rick Booth
'95 900SS SP
E-mail: rick@itron-ca.com
---

87. Message not deliverable

From: ADMINISTRATOR@MAF.Org
Date: Tue, 23 Jul 96 16:10:12 UTC
To: hfsig@tapr.org
Message-ID: <9606238381.AA838144057@gabriel.maf.org>
Rendered body type: text/plain

In a previous message, Phil Karn wrote:
>..... snip .....
>
> .... I've done this for a FFT (specifically a radix 4 FFT
> butterfly) and for a FIR.
>
>...... snip ....
>Phil
>
  How about sharing a code snippet with hfsig, showing the butterfly
  and the FIR ? The wheel has been reinvented so many times, let's
  avoid one more reinvention...   :-)

  Thanks in any case,

Alberto di Bene, I2PHD   dibene@vnet.ibm.com   i2phd@amsat.org
---

88. Message not deliverable

From: ADMINISTRATOR@MAF.Org
Date: Tue, 23 Jul 96 16:17:13 UTC
To: hfsig@tapr.org
Message-ID: <9606238381.AA838144149@gabriel.maf.org>
Rendered body type: text/plain

At 07:51 PM 7/22/96 -0500, you wrote:
>>I was talking to the developers of Pactor II the other week and they are
>>not willing to make the Pactor II specs available freely. I really don't
>>know any reason for this, but it is their decision. This means - at least
>>for German amateurs - that it could be illegal to use Pactor II.
>
>This policy (along with the similar proprietary attitudes of the companies
>making Clover and G-TOR) gives us an opportunity to create a new and open
>modulation and coding format more in the cooperative spirit of ham radio.
>And I believe it could easily outperform these proprietary schemes as well.
>
>Phil
>
           YES, YES, YES!!!!!!!!!!!

Outperform, outsell, obsolete them!

(Especially the guys who use paying hams to test [at a high price] stuff they
want to sell to the military!)

73, Jim
wb4gcs@amsat.org

89. Message not deliverable

From: ADMINISTRATOR@MAF.Org
Date: Tue, 23 Jul 96 16:16:16 UTC
To: hfsig@tapr.org
Message-ID: <9606238381.AA838144142@gabriel.maf.org>
Rendered body type: text/plain

>But you didn't expend the resources to develop the algorithm. SCS tells me
>they spent a very long time researching and testing the protocol. When you
>do that, you can't afford to give the final result away. I'd love to
>implement Pactor II myself, but I can understand why licenses aren't being
>offered cheap.

Keeping the protocol proprietary isn't necessarily going to pay back
their development costs either. The computer and communications
industry has a long history of proprietary technologies being quickly
displaced by more open, standard systems. Anybody here still use a
Telebit Trailblazer modem in PEP mode?

More enlightened companies are content to have part of something
rather than 100% of nothing.

Phil

90. Message not deliverable

From: ADMINISTRATOR@MAF.Org
Date: Tue, 23 Jul 96 16:18:59 UTC
To: hfsig@tapr.org
Message-ID: <9606238381.AA838144157@gabriel.maf.org>
Rendered body type: text/plain

> >>I was talking to the developers of Pactor II the other week and they are
> >>not willing to make the Pactor II specs available freely. I really don't
> >>know any reason for this, but it is their decision. This means - at least
> >>for German amateurs - that it could be illegal to use Pactor II.
> >
> >This policy (along with the similar proprietary attitudes of the companies
> >making Clover and G-TOR) gives us an opportunity to create a new and open
> >modulation and coding format more in the cooperative spirit of ham radio.
> >And I believe it could easily outperform these proprietary schemes as well.
> >
> >Phil
> >
>   YES, YES, YES!!!!!!!!!!!
>
> Outperform, outsell, obsolete them!
>
> (Especially the guys who use paying hams to test [at a high price] stuff they
> want to sell to the military!)

Unfortunately, it is even worse than selling just to the military: they sell /
trying to sell it to the humanitarian aid community (Red Crosses/Crescents,
United Nations agencies, non- and government organisations) - and at much
higher prices. Though the problems is, that there are currently no decent
alternatives really, besides SITOR/AMTOR or maybe GTOR/Pactor-1.

Rolf, HB9CWP

91. Message not deliverable

From: ADMINISTRATOR@MAF.Org
Date: Tue, 23 Jul 96 16:18:37 UTC
To: hfsig@tapr.org
Message-ID: <9606238381.AA838144153@gabriel.maf.org>
Rendered body type: text/plain

I agree Phil.

The issue was that these where what the FCC termed as being enough to allow
legal operations under the unspecified digital modes area.

Was not what the FCC agreed to the position that Paul Rinaldo and the Future
Systems Committee work towards ?

But as you point out -- these were not technical specifications.

There is nothing keep this group or amateur radio as a whole from developing
a standard that is more open and works better than current systems.

Greg

>
> >The ARRL has a publication that discusses the Clover, Pactor and GTOR.
> >These were presented at the 13th DCC I believe -- which was when the FCC
> >made the statement about publishing a standard to make it legal
>
> These papers discuss the general approaches used in each scheme, but they

> are by no means complete enough to write a compatible modem. So I certainly
> wouldn't call them "standards" or "specifications".
>
> Phil
>
>

92. Message not deliverable

From: ADMINISTRATOR@MAF.Org
Date: Tue, 23 Jul 96 16:15:47 UTC
To: hfsig@tapr.org
Message-ID: <9606238381.AA838144139@gabriel.maf.org>
Rendered body type: text/plain

>I was talking to the developers of Pactor II the other week and they are
>not willing to make the Pactor II specs available freely. I really don't
>know any reason for this, but it is their decision. This means - at least
>for German amateurs - that it could be illegal to use Pactor II.

This policy (along with the similar proprietary attitudes of the companies
making Clover and G-TOR) gives us an opportunity to create a new and open
modulation and coding format more in the cooperative spirit of ham radio.
And I believe it could easily outperform these proprietary schemes as well.

Phil

93. Message not deliverable

From: ADMINISTRATOR@MAF.Org
Date: Tue, 23 Jul 96 16:16:51 UTC
To: hfsig@tapr.org
Message-ID: <9606238381.AA838144146@gabriel.maf.org>
Rendered body type: text/plain

>The ARRL has a publication that discusses the Clover, Pactor and GTOR.
>These were presented at the 13th DCC I believe -- which was when the FCC
>made the statement about publishing a standard to make it legal

These papers discuss the general approaches used in each scheme, but they
are by no means complete enough to write a compatible modem. So I certainly
wouldn't call them "standards" or "specifications".

Phil

94. Message not deliverable

From: ADMINISTRATOR@MAF.Org
Date: Tue, 23 Jul 96 16:19:43 UTC
To: hfsig@tapr.org
Message-ID: <9606238381.AA838144168@gabriel.maf.org>
Rendered body type: text/plain

>The issue was that these where what the FCC termed as being enough to allow
>legal operations under the unspecified digital modes area.

I suspect that all the FCC really cares about is that they can get a box
to monitor the emission if they want to. That simply means it has to
be commercially available, which they all are.

Unfortunately, that does nothing to promote general appreciation and
understanding of what goes on in these newer modulation schemes. And I
find that deeply troubling, as it's very much against what I believe to be
the amateur radio spirit.

Phil

95. (no subject)

From: ADMINISTRATOR@MAF.Org
Date: Tue, 23 Jul 96 16:19:21 UTC
To: hfsig@tapr.org
Message-ID: <9606238381.AA838144164@gabriel.maf.org>
Rendered body type: text/plain

Subject: Message not deliverable

> How about sharing a code snippet with hfsig, showing the butterfly
> and the FIR ? The wheel has been reinvented so many times, let's
> avoid one more reinvention...   :-)

Okay, I've put the FIR filter on my web page:

http://www.qualcomm.com/people/pkarn/ham.html

I have to clean up and package the FFT code. When I do, I'll add it to the
same page.

Phil

96. Re: [HFSIG:1334] RE: HF Modem Bandwidth

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 23 Jul 1996 11:01:12 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607231801.LAA14798@servo.qualcomm.com>
Rendered body type: text/plain

>My guess is the biggest stumbling block to wideband com at HF is the
>base of transceivers already in place among the ham world...this puts
>the upper limit at about 2.4 KHz.

Well, mostly yes. But you can actually do a lot in 2.4 KHz, at least compared
to 500 Hz. And there's the possibility of frequency hopping the whole thing,
depending on the speed of the synthesizer and the computer control interface.

But I agree, wider filters are ultimately necessary. It may be
possible to modify many radios to provide them, although they are
really not widely (!) available. At Dayton I asked the Fox-Tango
people and they said they had never gotten a request for a *wider*
filter before...

Phil

97. Re: [HFSIG:1342] Message not deliverable

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Tue, 23 Jul 1996 18:00:13 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199607232300.SAA06286@www.kelly-afb.org>
Rendered body type: text/plain

Phil,

I've had conversations with some FCC Field Types and get the same feeling...
if they can obtain a "box" to monitor the "data", they are happy.

In a previous message you referenced RX bandwidth...this worrys me
because I feel that your/our effort shold be originally to work within
the bandpass of the xceivers on the air today with a built-in design to
go with wider bandwidths as xceivers (RF equipment) becomes available.

Your comments on PACTOR, G-tor and Clover are right on line. You
(plural)/we should build a better protocol and implement it on a DSP
sound card/one of the /dsp platforms or the TAPR platform whichever makes
the most sense for the protocol.

Walt on vacation in ID...hello Johan...was near your QTH yesterday
afternoon...Ok...about 150 miles away.

73 all.

>
> >The issue was that these where what the FCC termed as being enough to allow
> >legal operations under the unspecified digital modes area.
>
> I suspect that all the FCC really cares about is that they can get a box
> to monitor the emission if they want to. That simply means it has to
> be commercially available, which they all are.
>
> Unfortunately, that does nothing to promote general appreciation and
> understanding of what goes on in these newer modulation schemes. And I
> find that deeply troubling, as it's very much against what I believe to be
> the amateur radio spirit.
>
> Phil
>
>
>

--
==========

```
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development.      |
=======================================================================
|                                 |                                   |
|                                 | The greatest dangers to liberty   |
| Walt DuBose - K5YFW             | lurk in insidious encroachment    |
| E-Mail k5yfw@www.kelly-afb.org  | by men of zeal, well-meaning      |
| Business Telephone: (210)925-6081 | but without understanding.      |
|     Home Telephone: (210)696-3196 |                                 |
|                                 |   - Justice Louis D. Brandeis     |
|                                 |                                   |
=======================================================================
```

98. Re: [HFSIG:1345] Re: Message not deliverable

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 23 Jul 1996 19:46:00 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607240246.TAA16616@servo.qualcomm.com>
Rendered body type: text/plain

>In a previous message you referenced RX bandwidth...this worrys me
>because I feel that your/our effort shold be originally to work within
>the bandpass of the xceivers on the air today with a built-in design to
>go with wider bandwidths as xceivers (RF equipment) becomes available.

Well, I agree that at first we should strive for something that fits
in a SSB bandwidth, just to be able to use existing transceivers and
filters. So the problem is mainly with the FCC rules, insofar as they
limit us to 300 baud and/or 1Khz "mark/space shift" (not sure how that
applies to M-ary FSK). And only certain codes seem to be allowed.

And I also agree that our problems are more likely to be with the ham
population, not the FCC. That's certainly the case so far with the
proposed spread spectrum rule changes -- and these changes are really all
part of the same thing.

Phil

99. Re: HF bandwidth

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 23 Jul 1996 22:19:21 -0800
To: hfsig@tapr.org
Message-ID: <199607240532.WAA07907@PEAK.ORG>
Rendered body type: text/plain

Phil,

I changed the topic to get everyone confused.

>>In a previous message you referenced RX bandwidth...this worrys me
>>because I feel that your/our effort shold be originally to work within
>>the bandpass of the xceivers on the air today with a built-in design to
>>go with wider bandwidths as xceivers (RF equipment) becomes available.

Arn't we are getting closer to a new era of all-digital radios? That should
perhaps allow us to customise bandwiths as needed.

>
>Well, I agree that at first we should strive for something that fits
>in a SSB bandwidth, just to be able to use existing transceivers and
>filters.

I believe to do things in small steps and build confidence and enthuisiasm.
It will take some convincing putting a new type of emission in a slot that
has traditionally been dedicated for another purpose. I do, however, see no
problem with the new OFDM, FDM schemes, or even your new scheme being
introduced. At least the signals that that I have casually listened to,
sounds pretty much like a regular 300 baud HF packet signal. I would'nt be
surprised at all if folks would use their HF packet TNC's to try decode it.

>So the problem is mainly with the FCC rules, insofar as they
>limit us to 300 baud and/or 1Khz "mark/space shift" (not sure how that
>applies to M-ary FSK). And only certain codes seem to be allowed.

I thought that we posed that question to Paul Rinaldo at the last DCC. He
and Chris Imley looked into the matter. I was imformed by Chris, at Paul's
request, that they were sure that there was no problem using MFSK, i.e., the
multitone modems (I agree that part 97 is little fuzzy on that one). I
assume the 300 baud/1 kHz restrictions was taken into consideration. I do
think we are at liberty to take that for granted now. If we behave ourselves
and do a good job of using and promoting this technology, I can only hope
that it would find general acceptance.

>
>And I also agree that our problems are more likely to be with the ham
>population, not the FCC. That's certainly the case so far with the
>proposed spread spectrum rule changes -- and these changes are really all
>part of the same thing.
>
>Phil
>
>

Your'e probably right about this - the fear of the unknown, or giving up
some creature comforts.

--Johan

100. Re: [HFSIG:1346] Re: Message not deliverable

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Wed, 24 Jul 1996 00:51:54 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199607240551.AAA07490@www.kelly-afb.org>
Rendered body type: text/plain

Phil,

You guys build the protocol and make it work and I'll use it and die
trying to sell it to the ham community and get the FCC to let it be used.
I work for the government and selling good ideas to government officials
has been making me a living wage for 30+ years. I've only failed once and
that was with the Air Force Aeromedical Evacuation System but my idea has
now been taken up by the DEA, Navy, Marine Corp. and Coast Guard.

Walt

In your message you write:
>
> >In a previous message you referenced RX bandwidth...this worrys me
> >because I feel that your/our effort shold be originally to work within
> >the bandpass of the xceivers on the air today with a built-in design to
> >go with wider bandwidths as xceivers (RF equipment) becomes available.
>
> Well, I agree that at first we should strive for something that fits
> in a SSB bandwidth, just to be able to use existing transceivers and
> filters. So the problem is mainly with the FCC rules, insofar as they
> limit us to 300 baud and/or 1Khz "mark/space shift" (not sure how that
> applies to M-ary FSK). And only certain codes seem to be allowed.
>
> And I also agree that our problems are more likely to be with the ham
> population, not the FCC. That's certainly the case so far with the
> proposed spread spectrum rule changes -- and these changes are really all
> part of the same thing.
>
> Phil
>
>

--

| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development. |
	The greatest dangers to liberty
Walt DuBose - K5YFW	lurk in insidious encroachment
E-Mail k5yfw@www.kelly-afb.org	by men of zeal, well-meaning
Business Telephone: (210)925-6081	but without understanding.
Home Telephone: (210)696-3196	
	- Justice Louis D. Brandeis

==========

101. Re: [HFSIG:1313] Alexander wrote..

From: Bob McGwier <n4hy@ccr-p.ida.org>
Date: Wed, 24 Jul 1996 02:02:03 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199607240602.CAA15033@growler.ccr-p.ida.org>
Rendered body type: text/plain

I am in possession of three articles about PACTOR II. It contains everything
except the details of the interleaving algorithm and the fallback/upgrade
algorithm. They are written by the PACTOR folks for publication. No,
I am not willing to make a thousand copies for people. It was published in
the Digital Digest I believe but I would have to look and I won't be able
to for at least three weeks (when I begin unpacking at our new residence,
we are moving a whole five blocks). Someone might check on this.

Bob

102. How to define a new protocol

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 23 Jul 1996 23:05:25 -0800
To: hfsig@tapr.org
Message-ID: <199607240618.XAA11860@PEAK.ORG>
Rendered body type: text/plain

Hi all,

I am looking for suggestions on how to define a protocol. Shall I follow the
way that the CCIR did theirs, i.e., a couple of state diagrams and notes? I
found it to be comprehensive enough to build your own code implementation
from it.

Seem to remember some time ago, I read somewhere, of the exsistance of a
formal language to do this kind of thing. As far as I remember it was a
structured definition language where one could travel down a tree to unfold
lower levels of more complex parts. But my memory is vague.

Thanks,

--Johan

103. Re: [HFSIG:1347] Re: HF bandwidth

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 24 Jul 1996 01:08:13 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607240808.BAA17329@servo.qualcomm.com>
Rendered body type: text/plain

>Arn't we are getting closer to a new era of all-digital radios? That should
>perhaps allow us to customise bandwiths as needed.

Well, they'll still have their limits. I want no limits. Or at least a
radio with a fairly wide IF *and* the ability to frequency hop reasonably fast.

>I thought that we posed that question to Paul Rinaldo at the last DCC. He
>and Chris Imley looked into the matter. I was imformed by Chris, at Paul's
>request, that they were sure that there was no problem using MFSK, i.e., the
>multitone modems (I agree that part 97 is little fuzzy on that one). I
>assume the 300 baud/1 kHz restrictions was taken into consideration. I do
>think we are at liberty to take that for granted now. If we behave ourselves
>and do a good job of using and promoting this technology, I can only hope
>that it would find general acceptance.

But it's precisely those 300 baud/1 KHz restrictions that I want to eliminate.
Eventually I want to spread over the whole band with frequency hopping, but
to start I'd like to do MFSK over SSB bandwidths.

Phil

104. Re: [HFSIG:1348] Re: Message not deliverable

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 24 Jul 1996 01:09:06 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607240809.BAA17345@servo.qualcomm.com>
Rendered body type: text/plain

>You guys build the protocol and make it work and I'll use it and die
>trying to sell it to the ham community and get the FCC to let it be used.

Okay, you're on. Don't underestimate the job you've taken on. Getting an
STA for ourselves will be fairly easy. Getting the rules changed to permit
general use will be the trick.

Phil

105. Re: [HFSIG:1350] How to define a new protocol

From: chbrain@dircon.co.uk (Charles Brain)
Date: Wed, 24 Jul 1996 09:19:14 +0100
To: hfsig@tapr.org
Message-ID: <199607240831.AA28142@felix.dircon.co.uk>
Rendered body type: text/plain

>Hi all,
>
>I am looking for suggestions on how to define a protocol. Shall I follow the
>way that the CCIR did theirs, i.e., a couple of state diagrams and notes? I
>found it to be comprehensive enough to build your own code implementation
>from it.
>
>Seem to remember some time ago, I read somewhere, of the exsistance of a
>formal language to do this kind of thing. As far as I remember it was a
>structured definition language where one could travel down a tree to unfold
>lower levels of more complex parts. But my memory is vague.
>
>
>Thanks,
>
>--Johan

Hello Johan,

As someone that has to implement protocols for a living I prefer either
a comprehensive set of State Diagrams or better still SDL (Z.100 I think).
There was a description of this in the
ARRL 7th Computer Networking Conference.

- Charles

106. Re: [HFSIG:1349] Re: Alexander wrote..

From: David Seeler <seeler@upei.ca>
Date: Wed, 24 Jul 1996 06:06:25 -0003
To: hfsig@tapr.org
Message-ID: <01I7FVNQWLK6J25QUK@upei.ca>
Rendered body type: text/plain

On 24 Jul 96 at 1:16, Bob McGwier wrote:

>
> I am in possession of three articles about PACTOR II. It contains everything
> except the details of the interleaving algorithm and the fallback/upgrade
> algorithm. They are written by the PACTOR folks for publication. No,
> I am not willing to make a thousand copies for people. It was published in
> the Digital Digest I believe but I would have to look and I won't be able
> to for at least three weeks (when I begin unpacking at our new residence,
> we are moving a whole five blocks). Someone might check on this.
>
> Bob
>

Check out the PActor II home page:

http://www.scs-ptc.com/index8.html

There is a number of articles about the methodology etc there -
including a suggestion that AEA has purchased a license for PActor II

>
Best Regards,
David Seeler, VY2DCS

107. Re: [HFSIG:1350] How to define a new protocol

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 24 Jul 1996 02:26:01 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607240926.CAA17580@servo.qualcomm.com>
Rendered body type: text/plain

>I am looking for suggestions on how to define a protocol. Shall I follow the
>way that the CCIR did theirs, i.e., a couple of state diagrams and notes? I

No, that's one of the *worst* ways.

Write your code. Make it work. Clean it up and release it as a
reference implementation. Also write a pseudo-code description that
follows it reasonably closely (could be done as comments on the actual
code).

That's how you aid somebody else in producing their own implementation,
which is after all the goal of writing a standard.

Phil

108. Re: [HFSIG:1351] Re: HF bandwidth

From: chbrain@dircon.co.uk (Charles Brain)
Date: Wed, 24 Jul 1996 11:34:21 +0100
To: hfsig@tapr.org
Message-ID: <199607241046.AA06443@felix.dircon.co.uk>
Rendered body type: text/plain

>Well, they'll still have their limits. I want no limits. Or at least a
>radio with a fairly wide IF *and* the ability to frequency hop reasonably fast.
>
>Phil
>
>
Hi Phil,
What sort of hop rates, many hops per symbol (ie fast) or many symbols
per hop (slow)?

I guess if you are talking MFSK you are looking at fast hopping and
non-coherent detection ?

I wonder what the 'settling' time of the DDS chips are (at reasonable cost).

I suppose a number of synthesisers could be used and then switch
between them to achieve the hop rate.

- Charles

109. Re: [HFSIG:1356] Re: HF bandwidth

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Wed, 24 Jul 1996 13:27:44 +0200
To: hfsig@tapr.org
Message-ID: <31F608B0.33590565@ife.ee.ethz.ch>
Rendered body type: text/plain

Charles Brain wrote:

> I wonder what the 'settling' time of the DDS chips are (at reasonable cost).
Almost instantaneos. That is within less than 20 cycles of the
DDS clock, which is typically 20 MHz or greater
(with the quite old AD DDS AD7008, as an example)
Cost is about 20$ (probably much less in larger quantities)

vy 73s de Tom
HB9JNX/AE4WA
--
Thomas (Tom) Sailer EMail: sailer@ife.ee.ethz.ch
Weinbergstrasse 76 Ham Radio: hb9jnx @ hb9w.che.eu
CH-8408 Winterthur Phone: ++41 52 222 32 81

110. Re: [HFSIG:1349] Re: Alexander wrote..

From: Johan Forre <forrerj@peak.org>
Date: Wed, 24 Jul 1996 08:29:29 -0700 (PDT)
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <Pine.SUN.3.91.960724082510.9548A-100000@kira>
Rendered body type: text/plain

Hi Bob,

Thanks for the kind offer, but I'm afraid that those articles from the
Digital Digest are somewhat incomplete. I don't even think you will be
able to recreate the waveform, for example how they spaced the waveforms,
and what measures are taken for crest factor control. I also am missing the
handshaking protocol without which it's almost impossible to make much
sense of it.

The most complete reference that I have seen is on SCS's web page. Even
that is not complete, but it's better than nothing at all.

--Johan

On Wed, 24 Jul 1996, Bob McGwier wrote:

>
> I am in possession of three articles about PACTOR II. It contains everything
> except the details of the interleaving algorithm and the fallback/upgrade
> algorithm. They are written by the PACTOR folks for publication. No,
> I am not willing to make a thousand copies for people. It was published in
> the Digital Digest I believe but I would have to look and I won't be able
> to for at least three weeks (when I begin unpacking at our new residence,
> we are moving a whole five blocks). Someone might check on this.
>
> Bob
>
>

111. Re: [HFSIG:1351] Re: HF bandwidth

From: Johan Forre <forrerj@peak.org>
Date: Wed, 24 Jul 1996 08:35:12 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.91.960724083021.9548B-100000@kira>
Rendered body type: text/plain

Hi Phil,

On Wed, 24 Jul 1996, Phil Karn wrote:

>........

> But it's precisely those 300 baud/1 KHz restrictions that I want to eliminate.
> Eventually I want to spread over the whole band with frequency hopping, but
> to start I'd like to do MFSK over SSB bandwidths.
>
> Phil
>
>

Ah - now you're talking. I would bet that if you can demonstrate such a
system and show it not to be the monster that folks believe it to be, you
may be surprised how it will be accepted.

I still think the time is ripe for folks to start developing all-digital
transcievers for these purposes, i.e., wider IF's with SS capabilities.

--Johan

112. Yet Another Free Maths Program

From: chbrain@dircon.co.uk (Charles Brain)
Date: Wed, 24 Jul 1996 17:45:00 +0100
To: hfsig@tapr.org
Message-ID: <199607241657.AA00768@felix.dircon.co.uk>
Rendered body type: text/plain

For those of you into manipulating all things mathematical try out the
shareware program produced by MathViews at ..

http://www.mathwizards.com/

By the way I assume no-one has seen a copy of the ECC book I was asking
about. I have ordered a copy, I will let you know if I think it is any good.

- Charles

113. [HFSIG:1359] Re: HF bandwidth

From: Robert.Glassey@nmp.nokia.com
Date: Wed, 24 Jul 96 18:02:55 +0100
To: forrerj@peak.org, hfsig@tapr.org
Message-ID: <H0000292021c57d9@MHS>
Rendered body type: text/plain

> On Wed, 24 Jul 1996, Phil Karn wrote:
>> But it's precisely those 300 baud/1 KHz restrictions that I want to
>> eliminate. Eventually I want to spread over the whole band with
>> frequency hopping, but to start I'd like to do MFSK over SSB
>> bandwidths.

> Ah - now you're talking. I would bet that if you can demonstrate such
> a system and show it not to be the monster that folks believe it to
> be, you may be surprised how it will be accepted.
>
> --Johan

It would be very interesting to see it demonstrated in reality. I have never seen anything to sufficently prove that it would work even in theory. Not wanting to be a spoil sport here, but it appears to me that the widespread objections are founded in solid laws of physics.

Consider this:

The noise floor on 20m is about -120 to -110 dBm in a 3kHz bandwidth using an isotropic antenna. 3 dB degradation can make or break a DX contact so interference must be -120dBm or lower.

Pathloss at 1km is 55dB for freespace, and probably around 70dB given real ground effects (into the same isotropic rx antenna, since only relative RX levels are important - unless you antenna is really bad).

Thus and interference must be less than -50dBm in 3Khz bandwidth at the transmitter. Over 300Khz, the total transmit power must be less than -30dBm, or 1 microwatt to ovoid interference to a DXer 1km away, regardless of the spreading method. (assuming power is evenly spread).

A narrow band mode can transmit easily 100W without causing interference, even to his neighbour 20 meters away. Thus the narrow band guy has a full **** 80 dB **** power advantage! and he dosn't have to deal with dozens of high power signals on the band that would interfere with the wide band guy. 80dB is a lot of anything however you look at it and you would be giving it up if you went for a whole band spreading mode. The narrow band mode can also operate 600 stations in 300kHz, with no interference, and then reuse frequencies with a 10dB capture effect.

How can you possibly turn this situation into an advantage?

... I'd really like to know, so far no one has been able to show me.

Cheers,

Rob G0VTQ

114. APT with the EVM (fwd)

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Wed, 24 Jul 1996 11:00:01 -0800
To: hfsig@tapr.org
Message-ID: <1DC7AF94303@frl.orst.edu>
Rendered body type: text/plain

Hi all,

Here is some good news for those wanting to play some more with the 56K EVM. I just received an e-mail from Jabi EA2ARU:

73's and enjoy,

--Johan

------- Forwarded Message Follows -------

I have uploaded to ftp.tapr.org:/tapr/SIG/hfsig/upload the file evmapt.zip

This code permits receiving APT transmissions with an EVM board and
JVFAX 7.1 It is based on a work of Craig Newell, VK4YEQ and has been
adapted to EVM56002 by Jabi Agirre EA2ARU.

It has been succesfully tested. We are open to sugestions and
comments.
Jabi Agirre EA2ARU
        govier02@spritel.es
        The real connaisseur

Eduardo Jacob EA2??? (got license, waiting for callsign)
        jtpjatae@bi.ehu.es
        edu@kender.es
        Work in this project: Packager, Readme composer, Internet
        interface, public relations...

Eduardo Jacob - Area de Ingenieria Telematica
Departamento de Electronica y Telecomunicaciones
ETSII y de IT   Tel: +34-(9)4-427 8055
UPV / EHU   Fax: +34-(9)4-441 4041
Alda Urquijo s/n   E-mail: jtpjatae@bi.ehu.es
E-48013 - Bilbao (Spain)   : 100021,2212 Compuserve

115. Re: [HFSIG:1360] Yet Another Free Maths Program

From: Johan Forre <forrerj@peak.org>
Date: Wed, 24 Jul 1996 11:23:15 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.91.960724111930.26831A-100000@kira>
Rendered body type: text/plain

On Wed, 24 Jul 1996, Charles Brain wrote:

>....
> By the way I assume no-one has seen a copy of the ECC book I was asking
> about. I have ordered a copy, I will let you know if I think it is any good.
>
> - Charles
>
>

I have ordered a copy and will probably have it by the end of the
week. Will let you know my thoughts. I own three other cookbook-style
books by the same author and although not very well organized, I must
admit that I have revisited all of them a number of times. The last one I
got was on digital filter design. It also contained a disk.

My two cents worth.

--Johan

116. Re: [HFSIG:1355] Re: How to define a new protocol

From: Johan Forre <forrerj@peak.org>
Date: Wed, 24 Jul 1996 11:57:05 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.91.960724114508.29586A-100000@kira>
Rendered body type: text/plain

Hi Phil,

On Wed, 24 Jul 1996, Phil Karn wrote:

> >I am looking for suggestions on how to define a protocol. Shall I follow the
> >way that the CCIR did theirs, i.e., a couple of state diagrams and notes? I
>
> No, that's one of the *worst* ways.
>
> Write your code. Make it work. Clean it up and release it as a
> reference implementation. Also write a pseudo-code description that
> follows it reasonably closely (could be done as comments on the actual
> code).
>
> That's how you aid somebody else in producing their own implementation,
> which is after all the goal of writing a standard.
>
> Phil
>
>

That would be fairly consise, I agree, but like a loner would work (perhaps great minds do work alone? - do they?). I am not sure how we would be able to open some topic related to some nitty-griity detail before investing time and effort in a real implementation. For example: I would like to provide for three modes in QUATOR:

Mode A: unconnected broadcast with full ECC

Mode B: Hybrid ARQ for keyboarding

Mode C: networked (CDMA ?)

First; is it a good idea, if not what would be objectionable. If it is acceptable, what would do we need to cater for in message headers. Perhaps some graphical representation such as a state diagram or flow diagram can get to the substance easier.

Suggestions ?

--Johan

117. Re: [HFSIG:1350] How to define a new protocol

From: zs6awk@global.co.za (Danie Brynard)
Date: Wed, 24 Jul 1996 22:08:33 +0200
To: hfsig@tapr.org
Message-ID: <199607250006.WAA17914@lin01.global.co.za>
Rendered body type: text/plain

Just a thought:

Johan I find those formal definitions difficult to understand. Why not implement something practical where one play while you learn about the new protocol ? For instance something that will run on a DSP56002EVM or perhaps a 100MHz Pentuim and then interested experimenters can write it over for other platforms. Or is this too amateurish ? :-)

danie

>Hi all,
>
>I am looking for suggestions on how to define a protocol. Shall I follow the
>way that the CCIR did theirs, i.e., a couple of state diagrams and notes? I
>found it to be comprehensive enough to build your own code implementation
>from it.
>
>Seem to remember some time ago, I read somewhere, of the exsistance of a
>formal language to do this kind of thing. As far as I remember it was a
>structured definition language where one could travel down a tree to unfold
>lower levels of more complex parts. But my memory is vague.
>
>
>Thanks,
>
>--Johan
>
>
>
>

118. Re: [HFSIG:1365] Re: How to define a new protocol

From: Johan Forre <forrerj@peak.org>
Date: Wed, 24 Jul 1996 14:08:07 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.91.960724133916.9845A-100000@kira>
Rendered body type: text/plain

On Wed, 24 Jul 1996, Danie Brynard wrote:

> Just a thought:
>
> Johan I find those formal definitions difficult to understand. Why not
> implement something practical where one play while you learn about the new
> protocol ? For instance something that will run on a DSP56002EVM or perhaps
> a 100MHz Pentuim and then interested experimenters can write it over for
> other platforms. Or is this too amateurish ? :-)
>

> danie
>

No, not a bad idea if you are in learning mode and the scope of things
remain fairly constrained.

However, real life protocol implementation details, such as all the
exceptions to the rule, and special cases, gets in the way and things
tend to get pretty thick and unwieldy real fast. For example, have a look
at my Pactor I implementation where have supplied both C and assembly
source code (available in the HFSIG "upload" area as TORLIB.ZIP). See if
you can make head or tail of the actual protocol from that :-). This was
not intentional, but evolution. This is a good example of what tend to
happen - in the push to get things done, too little effort done to document
things properly. Once it works, one loose interest in the details.

I sense that the protocol definition should be timeless, and preferably not
tied to any particular DSP or RISC architecture. This way it would remain
there for posterity and give folks the chance to use whatever platform
they feel comfortable with, i.e., sound card, EVM or Pentium.

Thanks for the feedback,

--Johan

119. Re: [HFSIG:1356] Re: HF bandwidth

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 24 Jul 1996 14:51:18 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607242151.OAA02722@servo.qualcomm.com>
Rendered body type: text/plain

>What sort of hop rates, many hops per symbol (ie fast) or many symbols
>per hop (slow)?

Fast hopping (more than one symbol per hop) entails noncoherent
combining losses, so that doesn't make a whole lot of sense unless
you're in a hostile anti-jam environment. One symbol per hop seems
like a reasonable maximum. If that's too fast, we could hop less
often, as long as the rate is still fast enough to exploit frequency
diversity and produce a uniform loading on the band.

>I guess if you are talking MFSK you are looking at fast hopping and
>non-coherent detection ?

Non-coherent detection, yes. Basically you can think of the whole band as
containing N tones. For each symbol, you pick a subset M out of N and then
transmit the one tone out of M that corresponds to the encoded data.

It doesn't really matter how you pick the subset M of N, so you might
as well group them together. That means you can use a IF bandwidth that's
smaller than the entire band and hop around with a DDS.

>I wonder what the 'settling' time of the DDS chips are (at reasonable cost).

A true DDS has no settling time -- it hops essentially instantaneously. Of
course, a hybrid DDS/PLL will behave differently.

120. Re: [HFSIG:1359] Re: HF bandwidth

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 24 Jul 1996 14:58:23 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607242158.OAA02751@servo.qualcomm.com>
Rendered body type: text/plain

>Ah - now you're talking. I would bet that if you can demonstrate such a
>system and show it not to be the monster that folks believe it to be, you
>may be surprised how it will be accepted.

Based on my experience so far promoting the ARRL proposal for more
liberal spread spectrum rules, I'm skeptical. Everyone keeps screaming
about "raising the noise floor", and they don't accept arguments that,
on average, the noise+interference floor would actually be *lower* on
a crowded band with power controlled spread spectrum than with our
present uncontrolled narrowband techniques.

Phil

121. Re: [HFSIG:1366] Re: How to define a new protocol

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 24 Jul 1996 15:55:55 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607242255.PAA09032@servo.qualcomm.com>
Rendered body type: text/plain

>I sense that the protocol definition should be timeless, and preferably not
>tied to any particular DSP or RISC architecture. This way it would remain
>there for posterity and give folks the chance to use whatever platform
>they feel comfortable with, i.e., sound card, EVM or Pentium.

That's why you can write the standard in pseudo-code, i.e., an English description that is still structured like a computer program.

The problem I have with logic diagrams, SDL diagrams, and conventional English prose descriptions of protocols (e.g., anything in the ITU-T X. series) is that they are many steps removed from actual implementations. In fact, the people who write them frequently have little or no actual programming experience. And these things are invariably replete with subtle errors and dead ends (special cases that are not handled at all). Or at least they're way behind the actual implementations.

A standard written in pseudo-code that is derived from an actual working implementation has many advantages. First, having been derived from a working implementation means it's much more likely to be correct and to cover all the special cases. Second, it's much closer to the form a programmer would want to implement in some other computer language. Programming is pretty much universal; the differences between computer languages are relatively minor compared to the similarities that are captured in pseudocode.

A good example of a fairly complex protocol specified in pseudo-code is the internet Transmission Control Protocol (TCP) in RFC-793. I wrote my own implementation in C that closely follows the pseudo-code, and I found it far easier than implementing X.25 LAPB from the spec.

Today, given the enormous popularity in C I would give serious consideration to making a well-commented portable C implementation the official description of a protocol. After all, a computer language is designed to be as unambiguous as possible.

Phil

122. Re: [HFSIG:1361] Re: HF bandwidth

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 24 Jul 1996 17:28:49 -0700 (PDT)
To: hfsig@tapr.org
Cc: karn@qualcomm.com
Message-ID: <199607250028.RAA09259@servo.qualcomm.com>
Rendered body type: text/plain

>It would be very interesting to see it demonstrated in reality. I have
>never seen anything to sufficently prove that it would work even in
>theory. Not wanting to be a spoil sport here, but it appears to me that
>the widespread objections are founded in solid laws of physics.

Not really. To the extent that they're based on any analyses at all, the widespread objects are based more in absolute worst-case analyses, without looking at the big picture.

>The noise floor on 20m is about -120 to -110 dBm in a 3kHz bandwidth
>using an isotropic antenna. 3 dB degradation can make or break a DX
>contact so interference must be -120dBm or lower.

How often is noise (by which I assume you mean natural background noise) the limiting factor on 20m? As compared to QRM?

>Pathloss at 1km is 55dB for freespace, and probably around 70dB given
>real ground effects (into the same isotropic rx antenna, since only
>relative RX levels are important - unless you antenna is really bad).

How many serious 20m DXers use isotropic antennas? Wouldn't it be more reasonable to consider the front-to-back and front-to-side ratios of typical 3-element yagis?

>Thus and interference must be less than -50dBm in 3Khz bandwidth at the
>transmitter. Over 300Khz, the total transmit power must be less than

>-30dBm, or 1 microwatt to ovoid interference to a DXer 1km away,
>regardless of the spreading method. (assuming power is evenly spread).

Ah, but it *does* depend on the spreading method. Only direct sequence spreads its power into a uniform, noiselike distribution. Consider instead a medium-rate M-ary FSK system like I propose. Pick a symbol duration on the order of 50ms (20 Hz). A 64-ary scheme would therefore require 64x20=1280 Hz at any given instant. Now let's frequency hop on every symbol over the 350 KHz of the 20m band. 350 Khz/1280 Hz ~= 273. With a hopping rate of 20 Hz, one would expect, on average, to hit every 1280 Hz channel once every 273/20 = 13.7 seconds or so. (If the hopping pattern were a linear frequency chirp, this would happen exactly periodically). For a narrower target (such as a 400 Hz CW filter, which I assume you'd be using if you're really seriously working such weak DX), the hit rate would be proportionately longer: once every 43.75 seconds. And each "hit" would last just 50ms.

How often do fades occur on 20m? Static crashes? Intermittent QRM from key-down VFO spotting, key clicks, SSB spillover and "didit-dit" queries? Less often than once per 45 seconds? How long do they last? Shorter than 50ms?

Keeping everything else constant, yes, the average interference power is the same for frequency hopping as for direct sequence. But I suspect that most narrowband operators would find strong QRM a small fraction of the time to be less objectionable than continuous, low-level QRM (continuous at least when transmitting).

It would help to study the behavior of typical HF transceivers when designing such a modulation scheme. How do their AGCs respond to very short bursts of a strong signal? How about their now-unused "woodpecker" filters, could they be used to blank a somewhat shorter symbol time altogether? Shorter M-ary symbols would probably be preferable anyway, to increase the user data rate.

Bear in mind that the commonly expressed objections to spread spectrum deal solely with the near-far problem. First of all, I just don't accept that near-far is unique to spread spectrum, as anybody who has ever lived next door to another active ham knows very well. Sine waves are absolutely perfectly orthogonal only in theory. Real filters aren't perfect, real power amplifiers produce intermodulation distortion, and real VCOs produce phase noise.

Second, there are other effective ways to mitigate the near-far problem beyond what I've already described, such as designating DX subbands and excluding them from the frequency hopping list. Having subbands for "DX", "local", "regional" and the like -- without regard to mode -- makes far more sense than our present scheme of dividing them up by emission mode. And that's not just because modern technology is hopelessly blurring the distinctions between "voice", "image" and "data".

And third, given the widespread use of spread spectrum, any occasional increase in QRM to a narrowband receiver from a nearby SS station will, on average, be more than made up for by a significant *decrease* in total interference from distant stations, thanks to the increased power efficiencies SS makes possible. As crowded as the HF bands are, it is rare to find an HF link truly limited by thermal or natural noise -- except perhaps when the band is closed. So postulating that limit in a SS QRM analysis is simply unrealistic. In practice, almost every real HF link is ultimately limited by the din of QRM from many distant stations reusing the same channel.

And that's what makes it in the direct interest of the narrowband user that power-controlled spread spectrum become as widespread (!) as possible, because that's the only way to reduce the total amount of RF power floating around as background QRM.

I understand that these arguments may seem counterintuitive. But they are nonetheless correct when you look at the whole picture instead of drawing conclusions from unrealistic and limited assumptions.

Phil

123. Re: [HFSIG:1364] Re: How to define a new protocol

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 24 Jul 1996 18:26:02 -0700 (PDT)
To: hfsig@tapr.org

Message-ID: <199607250126.SAA15628@servo.qualcomm.com>
Rendered body type: text/plain

>Mode A: unconnected broadcast with full ECC

>Mode B: Hybrid ARQ for keyboarding

>Mode C: networked (CDMA ?)

Looks good. I've soured somewhat on type II hybrid ARQ/FEC since writing my paper some years ago, because I've learned that in practice it is very difficult to coherently combine symbols over a wide span. Coherent PSK loops have difficulty recovering carrier phase and noncoherent M-ary FSK schemes have their threshold effects. And of course whatever sync mechanism you use has to be much more robust than the coding, or it all breaks down much as a damaged CD will cause the typical CD player to skip a track well before it reaches the error correcting limit of the dual-RS code.

So it's probably just as well to stick with a type I hybrid (i.e., punt on the frame combining) as long as you have some sort of adaptation mechanism where you can change the code and/or modulator symbol rates.

Phil

124. Re: [HFSIG:1333] HF Modem Bandwidth

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 24 Jul 1996 18:28:58 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607250128.SAA15642@servo.qualcomm.com>
Rendered body type: text/plain

>10 dB is a lot--how much wider are you thinking, Phil? Tell me you don't
>want all of 20 meters!

As much of it as I can get, except perhaps for small DX windows. In exchange, of course, I'll do automatic power control and implement a highly efficient modem that will seriously minimize my total RF power requirements.

>I think to go much beyond 500 Hz in the CW/DATA portion of the bands would
>add years to the introduction of a new protocol. The educational effort
>required would be enormous. I think you could probably get the FCC to go
>along--it's the general ham population that would react very negatively to
>any kind of wideband proposal. At the minimum I think wideband signals

You are probably right, but it's easier to change the rules than the laws of physics. So there's really no choice. You're absolutely right, the FCC is far more enlightened about these things, but they won't change the rules until the general population can be educated enough to not oppose it in knee-jerk fashion.

Phil

125. Re: [HFSIG:1370] Re: HF bandwidth

From: k6sti@n2.net (Brian Beezley)
Date: Wed, 24 Jul 1996 20:58:23 -0700
To: hfsig@tapr.org
Message-ID: <199607250358.UAA01497@ravel.n2.net>
Rendered body type: text/plain

Phil, I admire the spirited way in which you defend the idea of HF spread spectrum, but I think SS is incompatible with traditional narrowband modes. I believe separate SS and narrow-emission HF band segments are necessary. I'll comment below on specific points, but the gist of my argument is as follows: When using a narrowband mode in a crowded band, it's always possible to find a substantially clear spot, except perhaps during contest weekends. Therefore, you really are limited by atmospheric and cosmic noise, not other transmitters. Widespread adoption of SS likely would raise the noise floor of all clear spots on a band and cause random--if infrequent--QRM blips from nearby SS transmitters.

>>It would be very interesting to see it demonstrated in reality. I have
>>never seen anything to sufficently prove that it would work even in
>>theory. Not wanting to be a spoil sport here, but it appears to me that
>>the widespread objections are founded in solid laws of physics.

>
>Not really. To the extent that they're based on any analyses at all,
>the widespread objects are based more in absolute worst-case analyses,
>without looking at the big picture.
>
>>The noise floor on 20m is about -120 to -110 dBm in a 3kHz bandwidth
>>using an isotropic antenna. 3 dB degradation can make or break a DX
>>contact so interference must be -120dBm or lower.
>
>How often is noise (by which I assume you mean natural background
>noise) the limiting factor on 20m? As compared to QRM?

For me it's always the limiting factor. The way I use HF is to QSY whenever QRM occurs. I don't fight it--it's too easy to find a clear spot.

>
>>Pathloss at 1km is 55dB for freespace, and probably around 70dB given
>>real ground effects (into the same isotropic rx antenna, since only
>>relative RX levels are important - unless you antenna is really bad).
>
>How many serious 20m DXers use isotropic antennas? Wouldn't it be more
>reasonable to consider the front-to-back and front-to-side ratios of
>typical 3-element yagis?

You're right that nobody uses isotropic antennas, but lots of people use dipoles with low directivity. I use two at right angles myself. But even if I used a good 3-element Yagi with 60-degree beamwidth, statistically it would reduce the incidence of near-transmitter, narrowband QRM only by a factor of six. I believe that widespread adoption of SS on HF in the same subbands as current narrowband modes would make this factor insignificant.

>
>>Thus and interference must be less than -50dBm in 3Khz bandwidth at the
>>transmitter. Over 300Khz, the total transmit power must be less than
>>-30dBm, or 1 microwatt to ovoid interference to a DXer 1km away,
>>regardless of the spreading method. (assuming power is evenly spread).
>
>Ah, but it *does* depend on the spreading method. Only direct sequence
>spreads its power into a uniform, noiselike distribution. Consider
>instead a medium-rate M-ary FSK system like I propose. Pick a symbol
>duration on the order of 50ms (20 Hz). A 64-ary scheme would therefore
>require 64x20=1280 Hz at any given instant. Now let's frequency hop on
>every symbol over the 350 KHz of the 20m band. 350 Khz/1280 Hz ~=
>273. With a hopping rate of 20 Hz, one would expect, on average, to
>hit every 1280 Hz channel once every 273/20 = 13.7 seconds or so. (If
>the hopping pattern were a linear frequency chirp, this would happen
>exactly periodically). For a narrower target (such as a 400 Hz CW
>filter, which I assume you'd be using if you're really seriously
>working such weak DX), the hit rate would be proportionately longer:
>once every 43.75 seconds. And each "hit" would last just 50ms.

I wouldn't want to engage in an SSB conversation overlayed with 50 ms tone bursts unless they occurred less frequently than once every few minutes. To put this in perspective, a 50-ms tone burst is the length of a 24-wpm Morse code dot.

>
>How often do fades occur on 20m? Static crashes? Intermittent QRM
>from key-down VFO spotting, key clicks, SSB spillover and "didit-dit"
>queries? Less often than once per 45 seconds? How long do they last?
>Shorter than 50ms?

If SS and narrowband segments didn't overlap, it wouldn't be necesary to add unevadeable SS QRM to the HF woes listed above.

>
>Keeping everything else constant, yes, the average interference power
>is the same for frequency hopping as for direct sequence. But I
>suspect that most narrowband operators would find strong QRM a small
>fraction of the time to be less objectionable than continuous,
>low-level QRM (continuous at least when transmitting).

Personally, I wouldn't continue a QSO that had strong QRM a small fraction
of the time. Instead, I'd QSY to a clear spot. But with SS across the
band, these no longer would be available.

>
>It would help to study the behavior of typical HF transceivers when
>designing such a modulation scheme. How do their AGCs respond to very
>short bursts of a strong signal? How about their now-unused
>"woodpecker" filters, could they be used to blank a somewhat shorter
>symbol time altogether? Shorter M-ary symbols would probably be
>preferable anyway, to increase the user data rate.

I don't think AGC response matters. The SS signal still will be reproduced
at some level. I think SS bursts are too long to blank. Blanking relies on
the impulse response of the IF filter to bridge a brief blanking gap. This
doesn't work if the gap is long. (However, linear predictive coding could
easily bridge a voice gap of several tens of ms.)

>
>Bear in mind that the commonly expressed objections to spread spectrum
>deal solely with the near-far problem. First of all, I just don't
>accept that near-far is unique to spread spectrum, as anybody who has
>ever lived next door to another active ham knows very well. Sine waves
>are absolutely perfectly orthogonal only in theory. Real filters
>aren't perfect, real power amplifiers produce intermodulation
>distortion, and real VCOs produce phase noise.

In 40 years of operating HF, I've never encountered a strong local signal I
couldn't evade by QSYing down the band a ways (excepting, of course, the
AGCless Hallicrafters S-38D I used in my novice days that would block out
the entire 40-meter band when the kid on the next street keyed up.) But the

recourse of QSYing wouldn't be possible with SS across the band.

>
>Second, there are other effective ways to mitigate the near-far
>problem beyond what I've already described, such as designating DX
>subbands and excluding them from the frequency hopping list. Having
>subbands for "DX", "local", "regional" and the like -- without regard
>to mode -- makes far more sense than our present scheme of dividing
>them up by emission mode. And that's not just because modern
>technology is hopelessly blurring the distinctions between "voice",
>"image" and "data".

I think this is the solution to the compatibility problem.

Did you know that many years ago the FCC proposed segmenting HF subbands not
by emission but by occupied bandwidth? I thought this was a terrific idea!
The ARRL nixed it, presumably in response to widespread negative membership
reaction.

>
>And third, given the widespread use of spread spectrum, any occasional
>increase in QRM to a narrowband receiver from a nearby SS station

>will, on average, be more than made up for by a significant *decrease*
>in total interference from distant stations, thanks to the increased
>power efficiencies SS makes possible. As crowded as the HF bands are,
>it is rare to find an HF link truly limited by thermal or natural
>noise -- except perhaps when the band is closed. So postulating that
>limit in a SS QRM analysis is simply unrealistic. In practice, almost
>every real HF link is ultimately limited by the din of QRM from many
>distant stations reusing the same channel.

This paragraph doesn't at all reflect my HF experience.

>
>And that's what makes it in the direct interest of the narrowband user
>that power-controlled spread spectrum become as widespread (!) as
>possible, because that's the only way to reduce the total amount of RF
>power floating around as background QRM.

I don't experience low-level background QRM. I simply don't put up with it.

>
>I understand that these arguments may seem counterintuitive. But they
>are nonetheless correct when you look at the whole picture instead of
>drawing conclusions from unrealistic and limited assumptions.
>
>Phil
>
>
>

I vote for separate--but equal!--SS subbands.

73--Brian, K6STI
k6sti@n2.net

126. Re: [HFSIG:1352] Re: Message not deliverable

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 25 Jul 1996 00:27:46 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199607250527.AAA11342@www.kelly-afb.org>
Rendered body type: text/plain

Oh, I'm not but think I'll have lots of help...and I'm not
underestimating the amount of work/time you and others have put in on
this "project". Thanks for your effort.

Walt

In your message you write:
>
> >You guys build the protocol and make it work and I'll use it and die
> >trying to sell it to the ham community and get the FCC to let it be used.
>
> Okay, you're on. Don't underestimate the job you've taken on. Getting an
> STA for ourselves will be fairly easy. Getting the rules changed to permit
> general use will be the trick.
>
> Phil
>
>
>

--
==
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development. |
==
| | |

```
|                                 | The greatest dangers to liberty |
| Walt DuBose - K5YFW             | lurk in insidious encroachment  |
| E-Mail k5yfw@www.kelly-afb.org  | by men of zeal, well-meaning    |
| Business Telephone: (210)925-6081 | but without understanding.    |
|     Home Telephone: (210)696-3196 |                               |
|                                 |   - Justice Louis D. Brandeis   |
|                                 |                                 |
=====================================================================
```

127. Re: HF bandwith (and UHF)

From: Iztok Saje <s52d@s55tcp.ampr.org>
Date: Thu, 25 Jul 96 06:34:13 EDT
To: hfsig@tapr.org
Message-ID: <69028@s55tcp.ampr.org>
Rendered body type: text/plain

Few cents from real life:
In Slovenia we have users packet QRGs with Wide-bandwith FM radios.
1 W, 200 kHz wide, TXdelay 10 to 15 ms, robust manchester coding,
speeds up to 76.8 kbit/sec.
Experience: This packet is unnoticable to plastic radio users (normal FM
handies), if you are not really near the wbfm packet station.
It is not SSB, but plastic radios are also not tecnical miracles.
(BTW, this is happening on 70cm band).
I guess noise-blankers, AGCs etc can filter out occasional QRM from SS.
If woodpecker QRM can be avoided, then SS packet should not be a problem on
20 m band.
Best 73, gl de Iztok

S52D/WU2D @S50BOX.SVN.EU
Info abt S5 packet: http://lois.kud-fp.si/hamradio/

128. Use of block allocations on the HF band?

From: "Bergzen Hakan, KARL" <hakan.bergzen@enator.se>
Date: Thu, 25 Jul 1996 11:11:00 +0100
To: HFSIG at TAPR <hfsig@tapr.org>
Message-ID: <320252A6@noak>
Rendered body type: text/plain

Even though the following does not directly affect the amateur radio
community some of you migh find the following interesting considering the
ongoing discussion:

The discussion regarding the radio Regulations versus a more flexible use of
HF bandwidth by the introduction of various new techniques are very
interesting. I am currently engaged in a small working party within ITU-R
SG9 studying this topic.

For cost reasons etc. the next generation of professional HF networks will
operate either unmanned or with very little real-time operator interaction.
This has promoted the development of automatic HF systems. A number of
adaptive functions are embedded in these HF systems (the most common one
being the automatic selection of most suitable traffic frequency) and they
are all using moderate-to-high speed HF modems. These factors have the
advantage of achieving a dramatic increase in link availability. (In some
cases even competing with satellite).

Background info: As of today around 15000 ALE systems (operating in
accordance with US MIL-STD-188-141A/FED-STD-1045A) have been procured
world-wide (two-thirds outside the US). And that is just one of the adaptive
systems on the market (although probably the largest). Today these
autoamtic/adaptive systems use most frequencies on the so-called
non-interference basis (with regard to the notified fixed frequency users).

However, these systems need to contain fairly large frequency pools,
covering all sorts of propagation conditions (and interference situations)
in order to be fully automatic. Therefore the need for a revision of the
radio regulations concerning the band allocations. Instead of having the
ITU-R allocating specific frequencies to each user/service it ought to be
allowed to (if meeting certain to-be-decided conditions) use certain defined
frequency blocks spread across the entire HF band (similar to the amateur
radio bands, but much wider). Observe that the approach is not to make a
new ITU band allocation, merely to allow frequency agile systems to operate
on some/several of the already defined ITU band allocations thus forming a
large block allocation.

The Swedish approach in this working party is that all sorts of frequency
agile systems should be usable on these frequency blocks, thus not limiting
it to be just adaptive HF systems but also SS-HF and SS-DS systems. The

systems to be used must of course be possible to monitor (openly available standards or openly procurable) and some sort of network registration at ITU-R will probably be needed (to sort out interference complaints from other users etc).

The aim is to have some sort of recommendation ready to be brought to the table at WRC-97. Feel free to contact me if you have any thoughts or suggestions on what types of (technical) restrictions that ought to be incorporated in this scheme. And especially if you are aware of tests, reports etc that have studied interference effects. Hopefully this scheme ought to reduce the overall band interference levels without causing harmful interference to fixed frequency users/services within the block allocations (which also remains to be defined).

Hakan Bergzen (hakan.bergzen@enator.se)
SM7OKI

129. [HFSIG:1368] Re: HF bandwidth

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 25 Jul 96 10:34:23 +0100
To: hfsig@tapr.org
Message-ID: <H0000292021d84a7@MHS>
Rendered body type: text/plain

Phil wrote:
> Based on my experience so far promoting the ARRL proposal for more
> liberal spread spectrum rules, I'm skeptical. Everyone keeps screaming
> about "raising the noise floor", and they don't accept arguments that,
> on average, the noise+interference floor would actually be *lower* on
> a crowded band with power controlled spread spectrum than with our
> present uncontrolled narrowband techniques.

Sorry to be a pain Phil, but I havent seen any such arguments, I've only seen SS proponents making bold and unsupported claims while the laws of physics appear to dissagree with them.

The basic SS argument makes several invalid assumptions:

1/ The bands are so conjested that DX is impossible, and QRM is the limiting factor in amateur communications.

2/ That cooperative, effective power contorl is possible on a global scale, both technically (near far problems) and politically.

3/ That nobody else in your neighbourhood will be transmitting or receiving on the same band as you to anyone further than the distance between you and your neighbour. The clasic near-far problem.

4/ That all hams prefer local celular type communications to DX.

5/ That all hams want to operate only one mode. Namely SS.

I'm actually in favour of SS, but only where appropriate, ie above 2GHz, in exclusive SS bands maybe 10 to 100MHz wide, in a celular style high speed random access HAM-LAN digital network. I think you would find a lot of support for this, inspiring enthusiastic developemnt, but on HF you'll always be swimming against the current, fighting a loosing battle, where either you or ham radio in general must loose.

Sorry to be such a spoil sport, lets get back to more realistic HF modems and developing an open standard for the next generation of ham friendly HF data modems.

More on standards later....

Cheers,

Rob

130. Re: [HFSIG:1358] Re: Alexander wrote..

From: Bob McGwier <n4hy@ccr-p.ida.org>
Date: Thu, 25 Jul 1996 09:43:24 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199607251343.JAA22782@wahoo.ccr-p.ida.org>
Rendered body type: text/plain

There is enough detail in the articles for you to get the waveforms and how they handle crest factor problems (I believe they ignore it and allow

interleaving and FEC fix it). The handshake protocol is what I meant by
upgrade/downgrade channels.

Bob

131. Re: [HFSIG:1373] Re: HF bandwidth

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 25 Jul 1996 07:49:35 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607251449.HAA00935@servo.qualcomm.com>
Rendered body type: text/plain

I guess it's not surprising to encounter resistance on this. John
Costas wrote his paper "Poisson, Shannon and the Radio Amateur" way
back in 1959, and it was already obvious that it was going to be a
hard sell. The old ways due hard...

I have much more to say later, but I wanted to make one quick point.
The central limit theorem from mathematics says that the sum of a
sufficiently large number of independent signals approximates a
Gaussian distribution. The characteristics of the individual signals
is surprisingly irrelevant, as long as they're independent. Even more
surprising is just how quickly it kicks in; just a few independent
sources are enough to make things look very Gaussian.

As an aside, this explains why Gaussian noise is so common in nature. Most
natural sources of noise consist of a very large number of independent
processes that may or may not be Gaussian. A single drop of water splashing
on the beach doesn't sound very Gaussian, but the crashing surf does.

So my quick point is this: how do you *know* that your link is limited
by natural background noise? How can you distinguish that from the din
of QRM from many distant background stations?

Phil

132. RE: [HFSIG:1362] APT with the EVM (fwd)

From: "Jabi" <govier02@sarenet.es>
Date: Thu, 25 Jul 1996 18:26:38 +0200
To: <hfsig@tapr.org>
Message-ID: <199607251621.SAA02267@sarenet.es>
Rendered body type: text/plain

> De: Johan Forrer <FORRERJ@frl.orst.edu>
> A: hfsig@tapr.org
> Asunto: [HFSIG:1362] APT with the EVM (fwd)
> Fecha: miércoles 24 de julio de 1996 20:02
>
> Hi all,
>
>
> Here is some good news for those wanting to play some more with the
> 56K EVM. I just received an e-mail from Jabi EA2ARU:
>
> 73's and enjoy,
>
> --Johan
>
>
>
> ------- Forwarded Message Follows -------
>
> I have uploaded to ftp.tapr.org:/tapr/SIG/hfsig/upload the file
evmapt.zip
>
> This code permits receiving APT transmissions with an EVM board and
> JVFAX 7.1 It is based on a work of Craig Newell, VK4YEQ and has been
> adapted to EVM56002 by Jabi Agirre EA2ARU.
>
> It has been succesfully tested. We are open to sugestions and
> comments.
> Jabi Agirre EA2ARU
>   govier02@spritel.es
>   The real connaisseur
 Please change de address. e-mail correct is:
govier02@sarenet.es

Tnx and 73, de Jabi, EA2ARU.

> Eduardo Jacob EA2??? (got license, waiting for callsign)
>   jtpjatae@bi.ehu.es
>   edu@kender.es
>   Work in this project: Packager, Readme composer, Internet
>   interface, public relations...
>
> Eduardo Jacob - Area de Ingenieria Telematica
> Departamento de Electronica y Telecomunicaciones
> ETSII y de IT   Tel: +34-(9)4-427 8055
> UPV / EHU   Fax: +34-(9)4-441 4041
> Alda Urquijo s/n   E-mail: jtpjatae@bi.ehu.es
> E-48013 - Bilbao (Spain)   : 100021,2212 Compuserve
>

133. Re: [HFSIG:1379] Re: HF bandwidth

From: k6sti@n2.net (Brian Beezley)
Date: Thu, 25 Jul 1996 09:54:29 -0700
To: hfsig@tapr.org
Message-ID: <199607251654.JAA12650@ravel.n2.net>
Rendered body type: text/plain

>I guess it's not surprising to encounter resistance on this. John
>Costas wrote his paper "Poisson, Shannon and the Radio Amateur" way
>back in 1959, and it was already obvious that it was going to be a
>hard sell. The old ways due hard...
>
>I have much more to say later, but I wanted to make one quick point.
>The central limit theorem from mathematics says that the sum of a
>sufficiently large number of independent signals approximates a
>Gaussian distribution. The characteristics of the individual signals
>is surprisingly irrelevant, as long as they're independent. Even more
>surprising is just how quickly it kicks in; just a few independent
>sources are enough to make things look very Gaussian.
>
>As an aside, this explains why Gaussian noise is so common in nature. Most
>natural sources of noise consist of a very large number of independent
>processes that may or may not be Gaussian. A single drop of water splashing
>on the beach doesn't sound very Gaussian, but the crashing surf does.
>
>So my quick point is this: how do you *know* that your link is limited
>by natural background noise? How can you distinguish that from the din
>of QRM from many distant background stations?
>
>Phil
>
>
>

The background noise I hear doesn't increase when the band opens.

More than that, I think it takes the superposition of hundreds of amateur signals before the sound approaches that of white noise. I base this on my experience at the receiving end of big CW and SSB pileups. On only one occasion did a pileup ever begin to sound white. This was during a six-meter opening into Japan at the peak of the last cycle when I was the only US station being heard. I remember the incident well because I knew about the Central Limit Theorem at the time, had never heard it in action on the ham bands, but realized it probably explained why the pileup was beginning to sound like noise. A station in Guam played back the 6m frequency to me on 10 meters so I could hear what it sounded like close to Japan. From there it sounded like a huge pileup of individual SSB stations. Even though for a short while the pileup sound became somewhat noisy, I could still hear individual stations pop up and was able to work them. There was no way you could mistake the phenomenon for natural noise. I've never experienced a similar sound on HF, even while monitoring the massive pileups that occur when new countries come on the air for the first time. I'd be amazed if any of the noise I hear on HF is due to the superposition of weak stations. However, I think your conjecture is fascinating and I'm going to ask several DX-contest operators I know who have years of experience dealing with big pileups and see what they can tell me.

73--Brian, K6STI
k6sti@n2.net

134. Re: [HFSIG:1378] Re: Alexander wrote..

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 25 Jul 1996 10:31:15 -0800
To: hfsig@tapr.org
Message-ID: <199607251744.KAA24655@PEAK.ORG>
Rendered body type: text/plain

Hello Bob,

>
>There is enough detail in the articles for you to get the waveforms and
>how they handle crest factor problems (I believe they ignore it and allow
>interleaving and FEC fix it). The handshake protocol is what I meant by
>upgrade/downgrade channels.
>
>Bob
>
>

Yes - you are right of course about the basic quadrature sine/cosine components, the other thing that I mean is which bit/dibit/tribit to phase transition allocations were used and how bit/dibit/tribits are encoded from the bit stream. Any number of allocations would be possible. Is there any form of interleaving, for instance. If so, what depth. Without these basic details I have some difficulty to reproduce the waveform from a data stream. Perhaps you could share some insight.

I found the reference to the tone spacing; it's 200 Hz or 2*T. As as you said seems like nothing is done about crest factor control. I do remind folks to check the earlier discussion here on HFSIG regarding interchannel interference when dealing with this kind of pulse shaping.

Besides missing the handshaking protocol, what about the frame makeup? We also need to know what control information there is, where it is located, and how it's encoded.

I also am a little ignorant about the compression scheme - I know the basic principle, and implemented the Huffmann compression of Pactor I. That required a table, which SCS kindly provided. But with Pactor II, I'm not sure what goes. Is Psuedo-Markov compression a dynamic scheme, i.e., building the table on the fly, like LZW, or is it some proprietary scheme that again uses their own tables. Without knowing how this works, it may as well be encrypted data.

Pactor II is, in my opinion, a really advanced system, well thought out, and has a lot of potential. I must compliment the designers. I still am a bit puzzled by all the controversy, half truths, and ignorance of the real issues that is being said and propagated as facts. That, perhaps is human nature. On the other hand, it is great to see all the activities and excitement about digital technology - is'nt it great!

--Johan

135. Spread Spectrum on HF

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Thu, 25 Jul 1996 11:38:48 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <199607251838.AA222059928@polecat.sr.hp.com>
Rendered body type: text/plain

Phil Karn <karn@qualcomm.com> wrote:

>>The noise floor on 20m is about -120 to -110 dBm in a 3kHz bandwidth
>>using an isotropic antenna. 3 dB degradation can make or break a DX
>>contact so interference must be -120dBm or lower.
>
>How often is noise (by which I assume you mean natural background
>noise) the limiting factor on 20m? As compared to QRM? ...

>>Pathloss at 1km is 55dB for freespace, and probably around 70dB given
>>real ground effects (into the same isotropic rx antenna, since only
>>relative RX levels are important - unless you antenna is really bad).
>
>How many serious 20m DXers use isotropic antennas? Wouldn't it be more
>reasonable to consider the front-to-back and front-to-side ratios of
>typical 3-element yagis?

It doesn't matter. Look, a decent communications receiver has over
100 dB of dynamic range. Reduce that by 20 dB to account for atmospheric
noise and background interference. That means the receiver can handle
an out-of-band interfering signal 80 dB above the effective noise level.

A direct-sequence spread spectrum transmitter 300 kHz wide will have 1%
of its power in the receiver's 3 kHz bandwidth. So an interfering SS
signal can be only 20 dB above the noise level if it is not to cause
interference. That's 60 dB worse than for narrowband modes.

>Only direct sequence spreads its power into a uniform, noiselike
>distribution. ...
>With a hopping rate of 20 Hz, one would expect, on average, to
>hit every 1280 Hz channel once every 273/20 = 13.7 seconds or so. ...

That's assuming the receiver frequency is exactly aligned with the
transmitting channels. If the receiver straddles two transmit channels,
the hit rate is twice as often. It also assumes a receiver bandwidth
of 1280 Hz. Normal SSB receivers have about twice that bandwidth.
The hit rate would be more like once every 4 or 5 seconds, which I
personally would find very annoying. If there are several SS signals
on the band, the hit rate would be totally unacceptable.

>For a narrower target (such as a 400 Hz CW
>filter, which I assume you'd be using if you're really seriously
>working such weak DX), the hit rate would be proportionately longer:
>once every 43.75 seconds. ...

How do you figure? Even if the receiver bandwidth were 1 Hz, it would
still get hit once every 13.7 seconds. Narrower bandwidth does lower
the level of the interference, however.

>Bear in mind that the commonly expressed objections to spread spectrum
>deal solely with the near-far problem. First of all, I just don't
>accept that near-far is unique to spread spectrum, as anybody who has
>ever lived next door to another active ham knows very well. Sine waves
>are absolutely perfectly orthogonal only in theory. Real filters
>aren't perfect, real power amplifiers produce intermodulation
>distortion, and real VCOs produce phase noise.

That's all true, and one could argue whether the interference advantage
of narrow-band modes is really 60 dB or some other number, but it's
still a huge advantage. (Actually, I think the 80 dB effective receiver
dynamic range number I used is, if anything, conservative.)

>And third, given the widespread use of spread spectrum, any occasional
>increase in QRM to a narrowband receiver from a nearby SS station
>will, on average, be more than made up for by a significant *decrease*
>in total interference from distant stations, thanks to the increased
>power efficiencies SS makes possible. ...

Dynamic power control can be used just as easily with narrowband digital
modes as with broadband spread spectrum.

>In practice, almost
>every real HF link is ultimately limited by the din of QRM from many
>distant stations reusing the same channel. ...

>The central limit theorem from mathematics says that the sum of a
>sufficiently large number of independent signals approximates a
>Gaussian distribution. ...

>A single drop of water splashing
>on the beach doesn't sound very Gaussian, but the crashing surf does.

>So my quick point is this: how do you *know* that your link is limited
>by natural background noise? How can you distinguish that from the din
>of QRM from many distant background stations?

That's an interesting theory. You're saying that what we perceive as
atmospheric noise is in reality thousands of weak distant signals added
together. I admit that there have been times when I have heard thousands
of weak signals added together (in contests and in pileups), and it indeed
does eventually start to sound a lot like random noise. However, the ear
has no trouble distinguishing it from Gaussian noise.

In any case, it doesn't affect the conclusion of the argument. Whatever
the background noise level, a strong spread spectrum signal on 20 meters
will raise the effective noise something like 60 dB more than a narrowband
signal of the same power.

AL N1AL

136. [HFSIG:1370] Re: HF bandwidth

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 25 Jul 96 19:40:53 +0100
To: hfsig@tapr.org
Message-ID: <H0000292021d84b2@MHS>
Rendered body type: text/plain

Hi Phil,

Thanks for the reply. I'm not trying to flame here, but only to express the concerns I and many other hams have, and to try to look at the detail to address these concerns.

> How often is noise (by which I assume you mean natural background
> noise) the limiting factor on 20m? As compared to QRM?

Where DX is concerned QRM is not the limiting factor, since you can usually find a clear frequency. Reducing QRM is certianly a good aim,

and if the QRM'er were using less power or spreading wider, then that QRM would be lower, but if everyone spread wider then the local guys would be the problem, not the DX QRM - clasic near far problem. As my analysis shows its the local guys QRM that is the limiting factor.

One intresting point hidden in your comment is the effect of fading, which I would consider more significant than DX QRM. As I understand it spreading can reduce data lost to fading. Even using only 2.4kHz there appears to be an advantage here. Still there are a lot less 2.4kHz channels for local hams to use. ('local' stations must be on differenct frequencies if the 1 microwatt syndrome is to be ovoided) There may be sufficent bandwidth for local operators to use 2.4kHz channels, and this may well have the DX QRM reducing advantages you mention. And the more efficent the modulation the lower the average TX power, so the fewer 'local' stations there are to share channels. There does appear to be a limit however, based on propagation conditions, average DX distances, and number of opertors.

> >Pathloss at 1km is 55dB for freespace, and probably around 70dB given
> How many serious 20m DXers use isotropic antennas? Wouldn't it be more
> reasonable to consider the front-to-back and front-to-side ratios of
> typical 3-element yagis?

For simplicity of analysis isotropic antennas were assumed. Where directional antennas are used the average interference level from DX stations does decrease, however we are concerned about the local noise floor, and a directional antenna at best would only give 20dB rejection, and at worst could give several dB gain! In the local enviroment averages don't work. Worst case must be assumed here, since it is the worst case that will cause problems.

> >regardless of the spreading method. (assuming power is evenly
> >spread).
>
> Ah, but it *does* depend on the spreading method. Only direct sequence
> spreads its power into a uniform, noiselike distribution. Consider
> instead a medium-rate M-ary FSK system like I propose. Pick a symbol
> duration on the order of 50ms (20 Hz). A 64-ary scheme would therefore
> require 64x20=1280 Hz at any given instant. Now let's frequency hop on
> every symbol over the 350 KHz of the 20m band. 350 Khz/1280 Hz ~=
> 273. With a hopping rate of 20 Hz, one would expect, on average, to
> hit every 1280 Hz channel once every 273/20 = 13.7 seconds or so. (If
> the hopping pattern were a linear frequency chirp, this would happen
> exactly periodically). For a narrower target (such as a 400 Hz CW
> filter, which I assume you'd be using if you're really seriously
> working such weak DX), the hit rate would be proportionately longer:
> once every 43.75 seconds. And each "hit" would last just 50ms.
>
> How often do fades occur on 20m? Static crashes? Intermittent QRM
> from key-down VFO spotting, key clicks, SSB spillover and "didit-dit"
> queries? Less often than once per 45 seconds? How long do they last?
> Shorter than 50ms?
>
> Keeping everything else constant, yes, the average interference power
> is the same for frequency hopping as for direct sequence. But I
> suspect that most narrowband operators would find strong QRM a small
> fraction of the time to be less objectionable than continuous,
> low-level QRM (continuous at least when transmitting).

Some good points also. We appear agreed that direct sequence is

definately out. Hopping clicks are definatly more acceptable.

```
CW         44 seconds between  clicks with a good filter
1280data   14 seconds
SSB         7 seconds
```

Of course the more stations the more clicks. With power control & ECC (simply keeping power down to minimum usable levels) the strength of each user will be lower, say on avreage 10-20dB lower. When each user becomes a click, how strong and how many clicks will their be?

Say there's 1 other local station, with power control S9+30dB, 5 strong stations, S9 with power control 20 weaker stations, s1-s5 with power control. You are working a S1 signal since he has reduced his power to the minumum level. You are loosing about 10% of the data. No too bad, with ECC.

An SSB station would hear roughly a click a second, with a bit of faster background clicking. Not nice, but modern noise blankers can handle this type of noise exceptionally well.

As the bands become more conjested data loss become more significant, especially for modes with little error correction. RTTY, amtor, pactor, packet, would become useless. It may be better to stick to SSB only portions of the band.

> It would help to study the behavior of typical HF transceivers when
> designing such a modulation scheme. How do their AGCs respond to very
> short bursts of a strong signal? How about their now-unused
> "woodpecker" filters, could they be used to blank a somewhat shorter
> symbol time altogether? Shorter M-ary symbols would probably be
> preferable anyway, to increase the user data rate.

Interesting that you comapre this proposed mode to the 'woodpecker'. It took a massive international effort to get rid of the much hated woodpecker (and massive political changes), this is *very* dangourous territory. If hams use a woodpecker like mode, then the ham bands become fair game for the next generation of over the horizon, space weapon resistant radars.

> Bear in mind that the commonly expressed objections to spread spectrum
> deal solely with the near-far problem. First of all, I just don't
> accept that near-far is unique to spread spectrum, as anybody who has
> ever lived next door to another active ham knows very well. Sine waves
> are absolutely perfectly orthogonal only in theory. Real filters
> aren't perfect, real power amplifiers produce intermodulation
> distortion, and real VCOs produce phase noise.

Very true, however TX intermod is limited to a few kHz of the signal, and crystal filters have a lot more rejection than any amount of procesing gain will give you. The last time I had problems with a local transmitter was when I was using a 1968 vintage FTDX400, which had fairly poor first IF selectivity and a S9+60dB CW signal was able to modulate the AGC. This is many orders or magnitude better than SS systems, where 1 watt at 10 miles can render the entire band useless.

> Second, there are other effective ways to mitigate the near-far
> problem beyond what I've already described, such as designating DX
> subbands and excluding them from the frequency hopping list. Having
> subbands for "DX", "local", "regional" and the like -- without regard
> to mode -- makes far more sense than our present scheme of dividing
> them up by emission mode. And that's not just because modern
> technology is hopelessly blurring the distinctions between "voice",
> "image" and "data".

This would go some of the way, but if two DXers were active in the same area band segragation would not help. Some form of carrier sense would also help, but we all know how poorly that works on HF packet.

> And third, given the widespread use of spread spectrum, any occasional
> increase in QRM to a narrowband receiver from a nearby SS station
> will, on average, be more than made up for by a significant *decrease*
> in total interference from distant stations, thanks to the increased
> power efficiencies SS makes possible. As crowded as the HF bands are,
> it is rare to find an HF link truly limited by thermal or natural
> noise -- except perhaps when the band is closed. So postulating that
> limit in a SS QRM analysis is simply unrealistic. In practice, almost
> every real HF link is ultimately limited by the din of QRM from many
> distant stations reusing the same channel.

> And that's what makes it in the direct interest of the narrowband user
> that power-controlled spread spectrum become as widespread (!) as
> possible, because that's the only way to reduce the total amount of RF
> power floating around as background QRM.

I understand your arguments here, although I think 90% of the gain here
can come from automatic power control and robust modem protocols alone.
Ovoiding nasty near-far/woodpecker problems altogether, that would
really be an advantage to narrow mode users.

SUMMARY... ish

You have presented quite a few excelent points, furthering the
understanding of these techniques in at least the technical ham
community. They're much appreciated by all I'm sure. The first step in
promoting an understanding of these principles is open discussion,
questions and answers in technical forums like HFSIG. Many thanks!

I think we are both agreed that Direct Sequence spread spectrum is
inappropriate on HF. And I'll admit many of my objections were based on
the assumption of some form of even power spreading.

It think the single most important point you have made is the difference
between direct sequence and hopping. Namely that even strong pulse
interference can be removed with impulse noise blankers and a neighbours
SS system will not kill yours regardless of the power he runs (within
reason). This dramatically improves the situation, and practically
eliminates the local near far problem. Near-far may still be a problem
when the bands are conjested since the many nearer dx stations will
swamp the farther ones. Hmmm, but if eveyone is running low power, the
few long distance DXers can increase their power without seriously
effecting the nearer low power stations working shorter distances.
however if many people worked long DX then short DX stations would have
to increase their power to compensate. But hopping certianly is much
better than direct sequence due to the its non-linear nature.

However we now have 'woodpecker' type interference on the band. The more
stations the worse it gets. On SSB, noise blankers can remove the
interference, however older digital modes with no ECC will suffer
greatly due to lost bits, and rappidly reach the point where they are
unusable. One solution to this is to only spread over the SSB part of
the band.

Reducing average QRM is a very worthy aim. However much can be gained
here with automatic power control and more power efficent modulation and
coding schemes. Manual ovoidance of most QRM is possible.

Spreading is a double edged sword. It may allow more power efficent
systems through its ability to overcome fadeing, although you can
probably get significant gain with a small degree of spreading (2.4kHz
or a bit more) but after that we may be into deminishing returns here.

Spreading wider does reduce the rate of clicking, but the resulting
increase of stations in the same band will increase the clicking rate
again. If there is a gain here it will be in the more even distribution
of click strengths.

On the other hand a woodpecker effect will be strongly resented,
especially by home brew enthusiests who may resent having to build a
noise blanker with no net gain to themselves, so a few data users can
have their 'antisocial' mode. The woodpecker was a much hated
interference source and removing the woodpecker from the HF bands was a
major achievement. Going back to the bad old days would be strongly
opposed. Remember we are no longer talking of below the noise
interference, this is very real 24 hour s9+ woodpecker interference for
only a small benifit (over narrow spreading) for the few HF-SS data
operators, that everyone else must live with, in the name of 'less
interference'

So it seems hopping is vastly better than direct sequence but still
antisocial. And the aims of reduced QRM are well worthwhile, but the
question we should be asking is how much do we need to spread to get a
significant QRM reduction, and what is the trade off with the annoyance
factor of the woodpecker effect?

Are narrow modes with power control sufficent? Will this reduce QRM
enough? The advantage here is we annoy nobody!

How about 2.4kHz spreads? How many extra dB's do we gain over narrow modes here. Is it worth taking up 5 times the bandwidth, in terms of other users and their rights to spectrum as a proportion of total users?

Can we go further and take say the top 50 or 100kHz and spread over that, how many toes will that step on? Is it feasible in a cooperative and friendly band sharing service like amateur radio? How many extra dB do we really gain here, in terms of QRM reduction? Is it worth it? Maybe if it really does work in practice this would be the way to start out, small at first without challanging to many people and making them turn defensive.

Does spreading over the entire band really have such a dramatic improvement that all amateurs should be forced to accept this new thinking, dramatically changing the outlook and practice of amateur radio forever? Can the amateur service not survive without it? Or is it so good that everyone will be falling over themselves to get onto this new mode to ovoid the QRM of the older modes, something like the AM to SSB switch? Even if this miricle does occur, surely operators must decide for themselves that the switch is what they want. Amateur radio is not a dictatorship. This makes a whole band (or most of the band) SS mode a thing of the distant future, with smaller SS sub-bands being the direction for now if even they are justified.

If it really does work, and if the international authorities can be convinced, maybe we could get a NEW BAND. This would keep everyone happy! Lots of good PR opertunities there, and it would encourage people to give it a go, without threatening existing modes and bands.

Sorry this is so long, but there are a lot of issues here and a lot to get my mind around.

Rob

137. Re: [HFSIG:1377] Re: HF bandwidth

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Thu, 25 Jul 1996 13:49:10 -0400
To: hfsig@tapr.org
Message-ID: <1F7C79C0.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

The only arguement I can find valid for having SS in the upper bands (UHF+) is for a wider spread bandwidth. I STILL haven't heard a valid arguement AGAINST SS only that so many people think that SS will "raise the noise floor." If every ham will use SS, maybe, but I contend this will never happen. I estimate the number of hams using this modulation scheme might grow to possibly 10%. And that's being liberal with the figures. Lets encourage SS experimentation, and IF interference occurs, THEN lets do something about it. Only until we have conclusive interference tests will this issue be put to rest.

73s de
Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1377] Re: HF bandwidth
Author: hfsig@tapr.org at BALT.SMTP
Date: 7/25/96 5:36 AM

Phil wrote:
> Based on my experience so far promoting the ARRL proposal for more
> liberal spread spectrum rules, I'm skeptical. Everyone keeps screaming
> about "raising the noise floor", and they don't accept arguments that,
> on average, the noise+interference floor would actually be *lower* on
> a crowded band with power controlled spread spectrum than with our
> present uncontrolled narrowband techniques.

Sorry to be a pain Phil, but I havent seen any such arguments, I've only seen SS proponents making bold and unsupported claims while the laws of physics appear to dissagree with them.

The basic SS argument makes several invalid assumptions:

1/ The bands are so conjested that DX is impossible, and QRM is the limiting factor in amateur communications.

2/ That cooperative, effective power contorl is possible on a global
scale, both technically (near far problems) and politically.

3/ That nobody else in your neighbourhood will be transmitting or
receiving on the same band as you to anyone further than the
distance between you and your neighbour. The clasic near-far
problem.

4/ That all hams prefer local celular type communications to DX.

5/ That all hams want to operate only one mode. Namely SS.

I'm actually in favour of SS, but only where appropriate, ie above 2GHz,
in exclusive SS bands maybe 10 to 100MHz wide, in a celular style high
speed random access HAM-LAN digital network. I think you would find a
lot of support for this, inspiring enthusiastic developemnt, but on HF
you'll always be swimming against the current, fighting a loosing
battle, where either you or ham radio in general must loose.

Sorry to be such a spoil sport, lets get back to more realistic HF
modems and developing an open standard for the next generation of ham
friendly HF data modems.

More on standards later....

Cheers,

Rob

138. [HFSIG:1381] Re: HF bandwidth

From: Kok Chen <kchen@apple.com>
Date: Thu, 25 Jul 1996 13:20:32 -0700
To: hfsig@tapr.org
Message-ID: <31F7D710.2081@apple.com>
Rendered body type: text/plain

Interesting discussion regarding spread spectrum.

I side with K6STI and a few others here in questioning whether
SS and narrow-band (compared to SS, even AM is narrow-band :-)
modes can coexist within the same sub-band.

One also has to be careful how one applies Central Limit Theorem,
and further, remembering that it is the probability density function
of a random process we are talking about and not its frequency
spectrum. So, just because crashing surf sounds white, it may well
not have Gaussian pdf.

A lot of us are probably familiar with a very common usage of
CLT, viz, that of adding 12 random numbers from a uniformly
distributed pdf, the result being an amazingly good approximation
of a random number with a Gaussian (normal) distribution function.
The magical 12 is especially pleasing since the resultant standard
deviation is precisely 1, starting with twelve unity width uniform
distributed random numbers.

But, wait a minute. Let's see what happens if we start with twelve
random numbers whose standard deviation ("RMS magnitude") are not
identical. Let's say one of the twelve is 10 dB "louder" than the
other eleven. The result is easy to visualise (since the pdf of
the sum of a set of random numbers is precisely the convolution
of the pdfs among themselves; no kidding, go look at the equations
involved) and is distinctly non-Gaussian. Indeed, the resultant
pdf will be shaped sort of like a raised rectangle, whose rising
and trailing edges are "gaussian" in "shape." What if the odd
signal is 20 dB "louder"? Gosh, the resultant pdf is almost the
same as the pdf of that loudest signal.

This makes the real world very ugly. Let's say there is a chap
who lives a few blocks from you. He runs SS and puts an S9+20dB
signal into your receiver. Forget all the white Gaussian noise
arguments, your evening of DX'ing is ruined (may your AGC [and
your ear drums] recover fast enough from that hypothetical 50ms
burst, HI).

The corollary is that if you have a neighbour like this, you will
need a zillion other S2 or S3 spread spectrum signals to be in the
same passband before that part of the sub-band obeys all the nice
theoretical properties. At the point when that happens, it doesn't

matter anyway; the theoreticians are now kept happy, but you will,
at that point, have an S9 noise from band edge to band edge!

The question is going to be, do I want to always have QRM in 10%
of all my QSOs, or have 1 out of 10 of my QSO's containing QRM.
Especially when, in the latter case, I could always QSY (well,
not if the other fellow is VK0WH, I guess :-) in the one out of
ten cases, and end up with no QRM.

73

Kok Chen, AA6TY kchen@apple.com
Apple Computer, Inc.

139. Noise Blankers

From: k6sti@n2.net (Brian Beezley)
Date: Thu, 25 Jul 1996 15:44:58 -0700
To: hfsig@tapr.org
Message-ID: <199607252244.PAA29512@ravel.n2.net>
Rendered body type: text/plain

>An SSB station would hear roughly a click a second, with a bit of faster
>background clicking. Not nice, but modern noise blankers can handle this
>type of noise exceptionally well.

I don't think any HF-modem scheme should rely on the use of common noise
blankers. As anyone who has had occasion to use one on a crowded HF band
knows, they are extremely susceptible to strong signals outside the narrow
IF filter passband but within the somewhat wider roofing filter passband.
The peaks of strong signals activate the blanker and devastate the received
audio. I use my TS-930S noise blanker only when absolutely necessary.
Often I'd like to engage it but can't.

Some VHF hams use homebrew blankers that sample RF outside the ham band in a
region with no signals. The blanker in one Collins HF receiver also sampled
noise pulses in the VHF range. But these kinds of blankers aren't common.

73--Brian, K6STI
k6sti@n2.net

140. Re: [HFSIG:1387] Noise Blankers

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 25 Jul 1996 21:41:46 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607260441.VAA24460@servo.qualcomm.com>
Rendered body type: text/plain

Whether a noise blanker helps or hurts a digital modem probably
depends on the specific situation. The most important thing is that
the receiver recover as quickly as possible after any strong burst of
interference, be it a wideband radar pulse or a relatively narrow
signal. What it does *during* the pulse is less important. And it is
very important that any NB not kick in when there is no pulse.

In a radio with a slow decay AGC, a noise blanker is a big help by
removing much of the pulse energy that would otherwise reach the AGC.
This is especially true with 70cm military radar.

An alternative is to just turn off the AGC so pulsed QRM can't affect
it. If dynamic range is a problem, let the demodulator control the
gain somehow, perhaps by driving a DAC wired back into the
radio. Since the demod has a better knowledge of what the desired
signal looks like than the radio does, it can do a better job of
setting the RX gain without being fooled by strong QRM.

Of course, to deal with impulse QRM, any modem will have to have good
coding and interleaving, as no blanker can actually restore what's
covered by a strong noise pulse.

What is normally thought of as the main function of a noise blanker,
muting the audio output during the noise pulse, is not so important
here. The modem can do this for itself if it has to. A hard-decision
FEC decoder probably won't care; the pulses simply cause errors that
the decoder has to correct.

A soft-decision Fano or Viterbi decoder has to take more care. You
don't want a strong noise pulse to mimic a data symbol with a very

strong (good) metric. You probably want to treat any burst of symbols
that's suddenly well above the average signal level as "outliers" and
erase them before decoding.

Phil

141. Re: [HFSIG:1386] Re: HF bandwidth

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 26 Jul 1996 01:53:25 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607260853.BAA25081@servo.qualcomm.com>
Rendered body type: text/plain

>One also has to be careful how one applies Central Limit Theorem,
>and further, remembering that it is the probability density function
>of a random process we are talking about and not its frequency
>spectrum. So, just because crashing surf sounds white, it may well
>not have Gaussian pdf.

Sure, the sum of a bunch of independent narrowband processes will
approximate a narrowband Gaussian. The linear summing process can't
create frequency components that weren't in the original signals. But
the result *will* look like noise, at least over a limited bandwidth.

In the real world of course, stations are scattered all over the band,
and many will overlap. So the ultimate limit in a very crowded band is
Gaussian noise that's bandlimited to the entire band.

And yes, you're right that if one source has a much larger variance
(power) than all the rest, it dominates the sum.

My point was just to get people thinking about the equivalence between
noise and interference. As a spread spectrum guy I already think that
way, but many of the same rules apply to ensembles of narrowband
signals.

You *have* to start thinking average and not worst case. If you insist
on the worst case game, then I could concoct a much worse worst case
for narrowband by assuming I put *all* of my power into your channel
as QRM, rather than just part of it. :-)

What matters in the end on a crowded band is the total interfering
signal power summed over all of the stations and not their individual
bandwidths. So to the extent that spread spectrum allows the use of
less average signal power by each station (thanks to FEC coding gain
plus the power savings from the extra frequency diversity in Rayleigh
fading), there will be less background QRM even for the narrowband users.

More later, I'm off to the Central States VHF Society conference in
a few hours, and I need sleep...

Phil

142. [HFSIG:1388] Re: Noise Blankers

From: Robert.Glassey@nmp.nokia.com
Date: Fri, 26 Jul 96 10:19:12 +0100
To: hfsig@tapr.org, karn@qualcomm.com
Message-ID: <H0000292021f32e2@MHS>
Rendered body type: text/plain

> An alternative is to just turn off the AGC so pulsed QRM can't affect
> it. If dynamic range is a problem, let the demodulator control the
Phil wrote:

> gain somehow, perhaps by driving a DAC wired back into the
> radio. Since the demod has a better knowledge of what the desired
> signal looks like than the radio does, it can do a better job of
> setting the RX gain without being fooled by strong QRM.

Yes, I agree. looking forward to when we have more digital receivers,
AGC controlled by the demodulator would be an effective solution for
these types of SS links. Error correction, which makes use of erasures
would effectivly 'smooth over' the pulses (as seem in the data stream).

The real problem is not the SS RX, but compatibility with the other
modes used on the band, ie digital modes with little ECC, and SSB.

Like I have mentioned, we cannot assume that all hams will want to use
only one mode. Forcing this would be the death nail in amateur radio.

Brian wrote:

> I don't think any HF-modem scheme should rely on the use of common
> noise blankers. As anyone who has had occasion to use one on a
> crowded HF band knows, they are extremely susceptible to strong
> signals outside the narrow IF filter passband but within the somewhat
> wider roofing filter passband. The peaks of strong signals activate
> the blanker and devastate the received audio. I use my TS-930S noise
> blanker only when absolutely necessary. Often I'd like to engage it
> but can't.
>
> Some VHF hams use homebrew blankers that sample RF outside the ham
> band in a region with no signals. The blanker in one Collins HF
> receiver also sampled noise pulses in the VHF range. But these kinds
> of blankers aren't common.

Like Brian says, and I'd forgotton, impulse noise blankers rely on the noise being wideband. A hopping signal may not trigger the noise blanker when required, and may trigger when it should not (ie the pulse is outside the final filter, but inside the roofing filter - with a much higher hit rate too! And of course under conjested conditions these noise blankers may not work anyway.

The 'Big Three' HF manufactureres would have to introduce a new spread spectrum pulse blanker, probably implimented in the the internal DSP units. Meanwhile the majority of hams using equipment of the 90's vintage and eariler would have to put up with it.

Rob, G0VTQ

143. Rob just to confirm: can you send me a copy of btl to: zs6awk@global.co.za ?

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Fri, 26 Jul 1996 13:32:12 +0200
To: hfsig@tapr.org
Message-ID: <s1f8caed.014@KIDD.CO.ZA>
Rendered body type: text/plain

Rob just to confirm: can you send me a copy of btl to: zs6awk@global.co.za ?

This other email of mine does not work well.

danie

144. List owner?

From: Mike Curtis <wd6ehr@kaiwan009.kaiwan.com>
Date: Sat, 27 Jul 1996 14:18:04 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199607272118.OAA23545@kaiwan009.kaiwan.com>
Rendered body type: text/plain

Who is the listowner? Because my address has changed slightly (from kaiwan.com to kaiwan009.com), I am unable to make changes in my account.

 -- mike

145. SCS business stratagy

From: Robert.Glassey@nmp.nokia.com
Date: Mon, 29 Jul 96 12:29:44 +0100
To: hfsig@tapr.org
Message-ID: <H0000292021eda1c@MHS>
Rendered body type: text/plain

Brian Wrote:
>> But you didn't expend the resources to develop the algorithm. SCS
>> tells me they spent a very long time researching and testing the
>> protocol. When you do that, you can't afford to give the final
>> result away. I'd love to implement Pactor II myself, but I can
>> understand why licenses aren't being offered cheap.

Phil wrote:
> Keeping the protocol proprietary isn't necessarily going to pay back
> their development costs either. The computer and communications
> industry has a long history of proprietary technologies being quickly
> displaced by more open, standard systems. Anybody here still use a
> Telebit Trailblazer modem in PEP mode?
>

> More enlightened companies are content to have part of something
> rather than 100% of nothing.

I have to agree with phil. The industry I'm in is based on open
standards and they are nessesary to allow the growth of this type of
technology, and of course the growth of our company. And IBM is still
one of the worlds largest suppliers of PC's!

Developing a standard is a lot of work of course, but expecting to make
the money back through selling a few licences at a high price may not be
wise or in the companies best interests. There's probably very little in
pactor II that we on HFSIG cannot duplicate without a licence. We may
not be allowed to make a compatible modem, but we could make one just as
good. If manufacturers, programers and ultimatly hams have a choice
between an expensive propietry protocol in very limited use and a free
open protocol thats just as good, then SCS may not be in a very good
position.

Delevoping (or being a major contributor) to a standard gives the
developer a head start on the competition and allows them to taylor the
protocol to suit them to some extent. It also allows them to stimulate
and grow at market that they will be in an ideal position to exploit.

The PTC II modem is very expensive, but then its a very complex bit of
kit, and would make most PCs it would be used with look pretty lame.
This box can 'pat its head and rub its belly at the same time while
wistling Waltz Sing Mytilder'. It seems aimed at commercial and
millitary users. Its not the sort of thing the average ham would have on
his bench, you may use it in a BBS, given its dual port abilities, but
how many hams would connect in pactor II. You'd bet better off getting
two PK232MBX's. If SCS is serious about PACTOR II in the amateur
enviroment is should make the standard open, or at least cheap,
otherwise it may never become popular. Even if it does, every year lost
is a years profits lost. SCS needs to bring out a cheap PACTOR II modem
to be competitive in an open market and to make a profit and cover its
developement costs though sales volumes. There is no reason why SCS
cannot produce a competitive DSP232 like box with PACTOR II, that could
become as much a standard as the PK232. This is a rapidly growing market
and they could take a fair share with an open standard and the right
products.

Alternativly a few 'real' hams could develop an open standard, or adopt
a MIL standard, and leave SCS in the lurch.

Lets develop an open standard and get better, faster and cheaper modems.

Rob

146. Re: [HFSIG:1393] SCS business stratagy

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Mon, 29 Jul 1996 10:32:45 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199607291532.KAA11287@www.kelly-afb.org>
Rendered body type: text/plain

BRAVO

In your message you write:
> Phil wrote:
> > Keeping the protocol proprietary isn't necessarily going to pay back
> > their development costs either. The computer and communications
> > industry has a long history of proprietary technologies being quickly
> > displaced by more open, standard systems.

                                                Anybody here still use a
> > Telebit Trailblazer modem in PEP mode?
        (Yes Phil but what's your point I know a guy running RTTY at 45 Baud
        with a tube converter and receiver/transmitter and he thinks this is
        state-of-the-art...a little humor helps the medicine taste better)

> I have to agree with phil. The industry I'm in is based on open
> standards and they are nessesary to allow the growth of this type of
> technology, and of course the growth of our company. And IBM is still
> one of the worlds largest suppliers of PC's!

        And 2X applies to hamradio or should...

>
> Developing a standard is a lot of work of course, but expecting to make

> the money back through selling a few licences at a high price may not be
> wise or in the companies best interests.
>   There's probably very little in
> pactor II that we on HFSIG cannot duplicate without a licence. [STUFF DELETED]
> but we could make one just as good.

        Very little indeed...I've worked with Mil-Std Modems on HF (I
        keep repeating myself) that run 2400 BPS on HF with little or
        no errors...expensive...yep initially at $56K, thne down to $15K,
        then down to $7K in 4 years. My bet is that the TAPR or Motorola
        or TI platforms and a 486 could do a darn good duplicating most of
        what they do...at least that's seems to be what I've been reading
        on this SIG for the past 2 years.

>

        If manufacturers, programers and ultimatly hams have a choice
> between an expensive propietry protocol in very limited use and a free
> open protocol thats just as good, then SCS may not be in a very good
> position.

        Its really more than just the protocol...its having hardware to
        make it "play". Remember the first TNCs? 8 1/2 X 11 boards and
        you used a VT-100 for the display and ran the beacon on and every 2
        minutes on 145.01. Then the TAPR TNC Kit...almost bought one when
        AEA came out with their first TNC.

        To sell it to the ham population in general you _must_ have
        available hardware for your software and it must be fairly easy
        to implement.

>
> The PTC II modem is very expensive, but then its a very complex bit of
> kit, and would make most PCs it would be used with look pretty lame.
> This box can 'pat its head and rub its belly at the same time while
> Alternativly a few 'real' hams could develop an open standard, or adopt
> a MIL standard, and leave SCS in the lurch.

        Don't forget a hardware platform if needed....

>
> Lets develop an open standard and get better, faster and cheaper modems.

        For the last 2.5 years that's what I thought that this SIG was
        all about.

>
> Rob
>
>

--
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development. |

	The greatest dangers to liberty
Walt DuBose - K5YFW	lurk in insidious encroachment
E-Mail k5yfw@www.kelly-afb.org	by men of zeal, well-meaning
Business Telephone: (210)925-6081	but without understanding.
Home Telephone: (210)696-3196	
	- Justice Louis D. Brandeis

147. LM1868 Application Note

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Mon, 29 Jul 1996 11:40:27 -0400
To: hfsig@tapr.org
Message-ID: <1FCDD080.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

I am trying to locate a source for the components in the National
     Semiconductor Application Note for the LM1868, particularly the

NR461EG, 1N60, and T3 (NS-107C made by Apollo Electronics).

I am also trying to locate a source for varactor diodes, preferably ranging from 140pF to 5pF. The idea is to replace the tuning capactor with varactors.

Can anyone help?

Robert

148. Re: [HFSIG:1322] ECC Book

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 30 Jul 1996 19:42:13 -0800
To: hfsig@tapr.org
Message-ID: <199607310254.TAA04021@PEAK.ORG>
Rendered body type: text/plain

Hi Charles,

I received my copy today and I think it is excellent.
The theory is gently introduced and the code examples may come in very handy. Certainly is worth putting on my shelf along with Lin & Costello, and Michealson & Levesque's books.

Thanks for pointing it out.

--Johan, KC7WW

(I ordered my copy from Powell's via the Internet (www.powells.com))

>Has anybody seen this book and if so is it worth adding to the 'library' ?.
>
>Error Coding Cookbook: Practical C/C++ Routines and Recipes for
> Error Detection and Correction
> Disk Included
>
> C. Britton Rorabaugh
>
>
> Hard ISBN 0-07-911720-1 1996
> 251p. 71 Illus. 7 3/8 x 9 1/4
> $55.00
>
>   ** Description **
>
> At last, the complexities of error coding are made simple If you
> don't happen to be an abstract mathematician, it's all too easy to
> get lost in the maze of error correction coding thats is a crucial
> component of any signal processing or communications system. Now
> this first-of-it-kind, non-theoretical guide clearly explains the
> often complex error coding process. You'll find complete
> step-by-step instructions on how to: Select the most appropriate
> code for your particular application; Design the corresponding
> coders and decoders; Implement many of the most well-known and
> frequently used codes for error correction. Plus, you'll have
> access to ready-to-use C and C++ programs for generating hundreds
> of codes for error detection and correction on the accompanying
> disk. The book's exceptionally clear descriptions of each code are
> reinformed by an array of worked-out examples and a brief review
> of only the necessary math where applicable. There has never been
> a more illuminating or practical treatment of this always
> challenging and occasionally perplexing area.
>
>   ** Audience **
>
> Electronics, electrical, communications, and computer engineers
> designing all types of communications and signal processing
> equipment.
>
>   ** Contents **
>
> Strategic Issues. Mathematical Tools for Coding. Block and Cyclic
> Codes. BCH and Reed-Solomon Codes. Encoders and Decoders.
> Convolutional Codes. Viterbi Decoding. Sequential Decoding.
>
>

149. Re: [HFSIG:1388] Re: Noise Blankers

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 31 Jul 1996 18:20:51 -0700 (PDT)
To: Robert.Glassey@nmp.nokia.com
Cc: hfsig@tapr.org
Message-ID: <199608010120.SAA09782@servo.qualcomm.com>
Rendered body type: text/plain

>these types of SS links. Error correction, which makes use of erasures
>would effectivly 'smooth over' the pulses (as seem in the data stream).

Error correction is an especially powerful technique with frequency hopped spread spectrum, as opposed to direct sequence. Both the frequency hopper and the narrowband user would benefit greatly from the ability to correct short bursts of errors, whether they occur from short bursts of interference or from natural channel degradations, such as static crashes and deep fades.

It's instructive to compare the error rate curves for digital modulation on fading channels with those for nonfading channels. On the nonfading additive white gaussian noise (AWGN) channel, the error rate of most modulation schemes decreases either exponentially with SNR (for noncoherent schemes like FSK and DBPSK) or according to the error-integral ("Q") function (for coherent BPSK).

But on a Rayleigh fading channel, both coherent and noncoherent schemes exhibit a error rate that decreases only *linearly* with increasing SNR. That is, if you want to decrease your bit error rate by a factor of 10, you need to increase your power by a factor of ten. The different schemes do differ in the precise amount of power required (e.g., ideal coherent BPSK is 3 dB better than DBPSK, which is 3 dB better than BFSK) but they are all exactly parallel over a very wide range of SNR.

Some illustrative figures: for a BER of 1%, uncoded BPSK on a Rayleigh channel requires an Eb/N0 of about 20 dB. To lower the BER to .1%, you need 30 dB. To achieve 10^-6, you need an coax-melting Eb/N0 of 60dB!

If you're after a low error rate on a fading channel, the proper use of coding and interleaving can have a dramatic effect. While coding can get you perhaps 7 dB at best on an AWGN channel, on a Rayleigh fading channel it's not hard to achieve as much as 35 - 40 dB in power savings compared to uncoded transmission. That's pretty impressive.

Now these gains are admittedly somewhat separate from the issue of spread spectrum frequency hopping. But some bandwidth expansion just from the coding is inevitable, and this expansion is likely to be greater than on the AWGN channel.

On the AWGN channel, you can get most of the achievable power gains with a bandwidth expansion of just 2-3x, e.g., by just adding rate 1/2 or rate 1/3 convolutional coding to coherent PSK. The extreme coding gains achieved on Galileo with concatenated codes require about 4-5x additional bandwidth.

But on the Rayleigh fading channel, both the achievable coding gains and the bandwidth expansion ratios are considerably greater. Not only are orthogonal signalling constellations with a fairly large number of dimensions necessary (e.g., M-ary FSK with large M), but the optimum FEC code rates for use with these schemes are lower than on the AWGN channel. For example, 16-ary FSK requires 4 times as much bandwidth as BFSK for the same data rate, and rate 1/8 coding is optimal for this value of M. That's a 32x bandwidth expansion, but it performs about 5 dB better than a rate 1/2 code combined with BFSK (which only doubles the uncoded bandwidth).

So my main concern with the existing rules is that they keep us from using modulation and coding methods that conserve RF power. The result is a larger average amount of interference power to both narrowband and wideband users. Rules that would let us use power-efficient coding and modulation would be good even if they did not allow spread-spectrum per se. But these techniques run in a continuum, and I'm not sure how one would make a formal distinction in the rules. (E.g., a rate 1/100 convolutionally encoded BPSK signal is hard to distinguish from one that is direct sequence spread with a process gain of 1:100).

Yes, it is possible to concoct worst cases where spread spectrum (or even a power-efficient wideband modulation scheme) could occasionally

interfere with a nearby narrowband user. But there are many ways to mitigate this problem. I've already described how frequency hopping can be used instead of direct sequence to reduce the subjective effect of a given amount of average interference. It's also quite possible with frequency hopping to watch for narrowband interference and drop it from the hopping sequence, benefitting both users. (The most general approach to adapting to nonuniform interference is known as "water filling", used in some OFDM modems).

And it may not always be necessary to spread at all; on a lightly loaded band, you could simply turn off hopping once you've found what appears to be an empty channel. (This is equivalent to dropping every channel but one from the hopping list).

But when the band gets loaded, and especially when the typical traffic consists of a lot of short, bursty transmissions, spread spectrum in general and frequency hopping in particular has the desirable effect of uniformly distributing the load over the band. The law of large numbers kicks in and the carrying capacity of the band increases thanks to the improved ability to dynamically share spectrum.

And this is ultimately why we should encourage spread spectrum on all of our bands, especially the most crowded ones.

Phil

August 1996

Original source file: hfsig.9608
Source: https://archive.org/details/tapr-1997
Recovered from: https://archive.org/download/TAPR-HF_1996-08/TAPR-HF_1996-08.txt

1. Advantages of SS

From: Gwyn Reedy <gwyn@paccomm.com>
Date: Thu, 1 Aug 1996 06:38:51 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB7F74.1B3654E0@gwyn.paccomm.com>
Rendered body type: text/plain

Phil Karn said (in conclusion to an excellent technical tutorial)

>(SS has) he desirable effect of uniformly distributing the load over the band. The law of large
>numbers kicks in and the carrying capacity of the band increases thanks to the improved ability to >dynamically share spectrum.

>And this is ultimately why we should encourage spread spectrum on all of our bands, especially >the most crowded ones.

I think this encapsulates the essence of SS resistance. Many amateur spectrum users (might I call out contesters and DXers) do not want to share spectrum; their satisfaction with the hobby derives from maximizing their personal communication effectiveness both on transmit and receive. Sharing is not one of their concerns. And this is the kind of person the ARRL considers the mainstream of amateur radio.

Gwyn Reedy, W1BEL
PacComm Packet Radio Systems
http://www.paccomm.com

---

2. Re: [HFSIG:1398] Advantages of SS

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 1 Aug 1996 05:51:32 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608011251.FAA18796@servo.qualcomm.com>
Rendered body type: text/plain

Gwyn's comments are perhaps a bit blunt, but he's certainly on the right track.

Another issue I've been meaning to comment on is the relative efficiency of relay vs direct communications. The near-far problem comes up mainly in the context of a station desiring to communicate to a very distant point, typically requiring high power, while nearby SS stations are communicating among themselves at relatively low power.

I believe Robert Glassey made an earlier comment along these lines when he said that SS communications assume local cellular-style communications. And he was quite right, if by "cellular style" he meant heavy geographical reuse of spectrum, with relaying rather than high power used to reach more distant stations. "Cellular style" does NOT mean you're unable to talk DX, you just may not be able to do it directly. (I routinely "work" hundreds of miles with my little 300mw handheld 890 MHz unit...)

This strikes at another major aspect of ham radio: the "self-sufficient ethic". Despite the literal meaning of the third letter in "ARRL" the amateur culture strongly favors direct end-to-end communications, especially in the HF bands. Repeaters are generally not allowed on these bands, and even where they're allowed (e.g., on 10m) they are usually excluded from most contests as they're considered to be "cheating".

Now some of this undoubtedly comes from the fact that historically, repeaters consume more spectrum than simplex and are also limited resources that are better reserved for other things, like mobile users and emergencies. But there is also an underlying theme of self-sufficiency that says a ham should have to do it all by himself to gain credit. Big antennas and massive linear amplifiers command a lot of respect, while resorting to a repeater is at least an admission of inferiority, or even downright dishonest (if used in a competition).

Spread spectrum turns this culture on its head. Not only does the
average ham have to accept the obviously absurd notion that a wideband
signal can be more spectrally efficient than a narrowband signal, but
he is also being asked to believe that repeaters can be more
spectrally efficient than simplex! That is, a network of relatively
low power store-and-forward spread spectrum packet radios with
min-energy routing is far more spectrally efficient than direct high
power end-to-end operation.

Even the reliability aspect of not wanting to depend on a repeater
(sometimes cited in the context of emergency communications) is really
not an issue here. All you need are enough stations to back each other
up and you can have a network that, overall, is much *more* reliable
than DX simplex thanks to the shorter inter-node propagation paths.

All this has actually always been true even for narrowband modulation;
the near-far aspect of spread spectrum just makes it more glaringly
obvious. With cooperative relaying, it would still be possible to
work DX on HF. You just wouldn't necessarily do it
directly. Unfortunately, for a lot of hams this wouldn't be the
same.

Phil

3. Shared SS

From: k6sti@n2.net (Brian Beezley)
Date: Thu, 1 Aug 1996 07:32:11 -0700
To: hfsig@tapr.org
Message-ID: <199608011432.HAA07242@ravel.n2.net>
Rendered body type: text/plain

>But when the band gets loaded, and especially when the typical traffic
>consists of a lot of short, bursty transmissions, spread spectrum in
>general and frequency hopping in particular has the desirable effect
>of uniformly distributing the load over the band. The law of large
>numbers kicks in and the carrying capacity of the band increases
>thanks to the improved ability to dynamically share spectrum.

Narrowband users already dynamically share spectrum. They QSY to avoid QRM.
But if their frequencies were overlayed with SS, they would experience QRM
they couldn't evade.

Any attempt to mix SS and narrowband modes on HF will have to address the
following complaint from day one:

Before this thing you call spread spectrum came along, I could always have a
nice chat on 20-meter SSB. If QRM arose, we either convinced the other guy
to move or we moved ourselves. But now, whenever I listen on the band I
hear funny noises in the background. I can still hold a conversation but
it's not pleasant listening to this stuff. I tried QSYing to find a clear
spot but it didn't help--the damn stuff is everywhere. I missed a new
country last week when that computer brat down the street fired up on SS and
took out the whole band for hours. When the SS guys across town get on, I
can hear well enough to check into the Old Farts net, but I often miss my
turn when net control fades into their bleeps and bloops. Frankly, I don't
get on 20 much any more because it's just not worth the trouble. I want my
band back.

Any technical argument for SS that can't respond simply and directly to a
complaint like this will be immediately dismissed as BS. Shared SS won't
happen until hundreds of thousands of hams like the one imagined above are
convinced that SS will bring benefit instead of destroying their present ham
activities.

Brian Beezley, K6STI
k6sti@n2.net

4. Re: [HFSIG:1400] Shared SS

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 1 Aug 1996 10:51:17 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608011551.KAA23596@www.kelly-afb.org>
Rendered body type: text/plain

In your message you write:
>
> Narrowband users already dynamically share spectrum. They QSY to avoid QRM.

> But if their frequencies were overlayed with SS, they would experience QRM
> they couldn't evade.
>
> Any attempt to mix SS and narrowband modes on HF will have to address the
> following complaint from day one:

        Way back there in my AFRes days, we were running SSB on several
        channels in the 8.4 - 8.9 MHz and in the 11.5 MHz range and the spooks
        about 300 meters were running SS between 3 -15 MHz to the same
        locations that we were. Propagation to the location we were both
        talking to was fair on 8 MHz and poor on 11 Mhz with our equipment.
        The spooks had very good communications. We were running 500 Watts
        PEP and they were running less that a 100 watts IAW their specs.
        Also, it didn't matter if they were on-the-air or not, it had no
        affect on our communications and I don't recall seeing an
        increase in channel noise...it darn sure didn't bother the marine
        CW station on the channel ajacant to our main channel (frequency).
        Also, there was another spook station located about 5 miles away
        and we couldn't tell when either was transmitting unless we called
        them on the fieldfone and asked. I might also mention that they
        were running broadband antennas and we were running dipoles so our
        antennas were better.

>
> Before this thing you call spread spectrum came along, I could always have a
> nice chat on 20-meter SSB. If QRM arose, we either convinced the other guy
> to move or we moved ourselves. But now, whenever I listen on the band I
> hear funny noises in the background. I can still hold a conversation but
> it's not pleasant listening to this stuff. I tried QSYing to find a clear
> spot but it didn't help--the damn stuff is everywhere. I missed a new
> country last week when that computer brat down the street fired up on SS and
> took out the whole band for hours. When the SS guys across town get on, I
> can hear well enough to check into the Old Farts net, but I often miss my
> turn when net control fades into their bleeps and bloops. Frankly, I don't
> get on 20 much any more because it's just not worth the trouble. I want my
> band back.
>
> Any technical argument for SS that can't respond simply and directly to a
> complaint like this will be immediately dismissed as BS. Shared SS won't
> happen until hundreds of thousands of hams like the one imagined above are
> convinced that SS will bring benefit instead of destroying their present ham
> activities.
>
>

        Based on my experience (see above) I have to disagree.

> Brian Beezley, K6STI
> k6sti@n2.net
>
>
        Walt/K5YFW

```
==========================================================================
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development.      |
==========================================================================
|                                  |                                   |
|                                  | The greatest dangers to liberty   |
| Walt DuBose - K5YFW              | lurk in insidious encroachment    |
| E-Mail k5yfw@www.kelly-afb.org   | by men of zeal, well-meaning      |
| Business Telephone: (210)925-6081 | but without understanding.       |
|     Home Telephone: (210)696-3196 |                                  |
|                                  |   - Justice Louis D. Brandeis     |
|                                  |                                   |
==========================================================================
```

5. Wide band modes

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 1 Aug 96 17:42:08 +0100
To: hfsig@tapr.org
Message-ID: <H000029202272ac5@MHS>
Rendered body type: text/plain

Hi Phil,

> Error correction is an especially powerful technique with frequency
> hopped spread spectrum, as opposed to direct sequence. Both the
[snip]

> But on a Rayleigh fading channel, both coherent and noncoherent
> schemes exhibit a error rate that decreases only *linearly* with
> increasing SNR. That is, if you want to decrease your bit error rate
> by a factor of 10, you need to increase your power by a factor of
> ten. The different schemes do differ in the precise amount of power
> required (e.g., ideal coherent BPSK is 3 dB better than DBPSK, which
> is 3 dB better than BFSK) but they are all exactly parallel over a
> very wide range of SNR.
>
> Some illustrative figures: for a BER of 1%, uncoded BPSK on a Rayleigh
> channel requires an Eb/N0 of about 20 dB. To lower the BER to .1%, you
> need 30 dB. To achieve 10^-6, you need an coax-melting Eb/N0 of 60dB!

> If you're after a low error rate on a fading channel, the proper use
> of coding and interleaving can have a dramatic effect. While coding
> can get you perhaps 7 dB at best on an AWGN channel, on a Rayleigh
> fading channel it's not hard to achieve as much as 35 - 40 dB in power
> savings compared to uncoded transmission. That's pretty impressive.

It is indeed. I have no arguments here. I've long suspected as much, although I've never had the numbers that show it. Although even very simple ECC can make back most of this 35-40db difference. ie and AMTOR ARQ works well with a BER of 10-2 (1%), as does RTTY where the operator fills in the errors. I think BPSK requires about 4 or 5dB for 1% BER in a AWGN channel, so were talking about 15dB extra power over the non-fading case. I guess most of this need for power is to give enough fade margin so that the signal only drops below the threshold 1% of the time.

I guess without any extra power, the signal would be above the threshold 50% of the time. Thus we have a 50% BER. A 1/2 rate ECC would ideally deal with this, although I guess a bit more redundancy would be usefull.

In the light of this 15dB fade margin, I wonder how much effect soft decoding would really have. Would erasure decoding be just as good? What I'm geting at is PACTOR II's K=9 or is it 11 contraint length. Would k=5 be just as good? Or could we just go for plain Reed Soloman coding? This would make PC implementation of an HFSIG protocol more realistic.

But getting back to your spreading argument, I've just suggested trippling the bandwidth to overcome fading, or have I?

I recall Ungerboch (I can never spell that) said that the solution was to go for multi levels, even if you cannot actually resolve each one discretly. Enough extra bits get through to give you the ECC bits for free (power and bandwidth-wise). This works in the AWGN channel, but in the fading channel it could be a real bonus, since some of the time the signal is several times what is required, allowing many additional levels. I imagine special attention would have to be paid to equalisation to overcome ISI when phase levels of 16+ PSK are used. (I've done on air measurements of PACTOR II transmissions using 8-psk, and the phase coherence is remarkably good over the DL-G path on 14MHz). Even amplitude levels could be considered since for a significant proportion of the time (>90%) they could be resolved. 64-QAM may not be unrealistic!! - giving an actual QPSK data rate, while overcomming the fading, and allowing high data rates when conditions are good. PACTOR II and clover seem to do the reverse, reducing ECC as level increase. Ungerboch says this is the wrong way around.

> Now these gains are admittedly somewhat separate from the issue of
> spread spectrum frequency hopping. But some bandwidth expansion just
> from the coding is inevitable, and this expansion is likely to be
> greater than on the AWGN channel.

Not inevitible, but I can still see that a wider band signal will have faster fades that are more easilly dealt with. I think the wider bandwidth helps when you consider interference from other users. Still a narrow a bandwidth allows more users in the same wide bandwidth without interference. 6 of one, half dozon of another as they say. I dont think either can make full use of the bandwidth.

> But on the Rayleigh fading channel, both the achievable coding gains
> and the bandwidth expansion ratios are considerably greater. Not only
> are orthogonal signalling constellations with a fairly large number of
> dimensions necessary (e.g., M-ary FSK with large M), but the optimum
> FEC code rates for use with these schemes are lower than on the AWGN
> channel. For example, 16-ary FSK requires 4 times as much bandwidth as
> BFSK for the same data rate, and rate 1/8 coding is optimal for this

> value of M. That's a 32x bandwidth expansion, but it performs about 5
> dB better than a rate 1/2 code combined with BFSK (which only doubles
> the uncoded bandwidth).

Hmm, this is interesting, expecially the 5dB increase over the BPSK
case. I wonder what it would be over the 64-QAM case.

So we are now talking about say 500Hz x 32 or about 16kHz bandwidth.
Interesting. Of course this is not hopping yet.

> And it may not always be necessary to spread at all; on a lightly
> loaded band, you could simply turn off hopping once you've found what
> appears to be an empty channel. (This is equivalent to dropping every
> channel but one from the hopping list).

How resistant would this M-ARY scheme be to other signals? I wonder if
different convolution codes could be used to provide further resistance
between signals (getting very spread spectrum like now), with some
sequential soft decoding being used to distinguish between the different
stations, rather than just using a majority vote detector. Could this
provide enough resistance to other stations on the same channel? It may
be possible to decode 100% a weaker station while a much stronger
station, or two use the same channel. MARY FSK seems to have quite an
advantage here.

Hopping inaddition to this would allow for more stations again.

I'm actually fairly well convinced that with a properly designed digital
radio, with a wide bandwidth and demod controlled AGC could overcome the
near far problem quite well, especially if spread enough so that local
impulses cause a BER of < 10%. I even like the idea!

The concern is about sharing the band with other users. I dont think
there is much concern about sharing the band with other wideband users,
and I can see that it may well be a more efficent way of using the
allocated spectrum, but sharing in an amateur enviroment is not just
about getting the maximum number of user into a particular band. Amateur
radio must allow many different modes, each with different strengths and
weakness to co-exist without causing each other interfence. The amateur
service is an 'experimental' service, which allows people to 'play' with
radio techniques and it many different forms to learn more about radio,
and have fun doing it.

If this mode can be made to work in a unspread (25kHz) bandwidth (same
25kHz allocation world wide) then I would support it, as a start, and if
it proves so popular that spreading becomes nessesary to accomodate the
number of stations, then there may be a good argument for allocating say
100KHz to the mode allowing hopping, without seriously effecting other
hams and their legitimate interests. If demand further increases then we
may have a good case for a NEW BAND, especially as HF spectrum is being
slowly cleared of commercial users.

I would support all these cases, but I could never support turning over
whole existing bands to spread spectrum. Its not in the spirit of ham
radio.

I posted a list of various assumptions that are made when arguing the
case for spread spectrum. Most of them still stand, although I think we
have improved the near-far problems.

1 & 4. Hams do want to work DX, simplex. Its not selfish, its the spirit
of ham radio, its what excites many people about the hobby. Maybe when
the bands are so badly conjested that DX is impossible SS may be our
only hope, but that would reduce ham radio to a single mode, with little
advantage (if any) over commercial celular data networks and the
internet. I think that day would be an especially black one for the
furure of the hobby, and many would feel ham radio is not worthwhile
anymore.

2 & 3. The near far thing. Still a problem for non-fec data users, and
SSB operators without DSP based noise blankers. They still have the
right to experiment with less than perfect modes, everyone must start
somewhere, and simple modes are a good starting point.

5. Ham radio is an experimental service for self education. Reducing ham
radio to a single mode, with only the prospect of contact with your
local ISP, or someone only a few 100 miles away being considered DX,
would pretty much spell the end of amateur radio. If we can only use it
as the poor mans data network, then the rich men will get it.

Maybe this is the future of ham radio, but I'd like to think otherwise.

Rob.

6. Re: [HFSIG:1400] Shared SS

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Thu, 1 Aug 1996 13:38:36 -0400
To: hfsig@tapr.org
Message-ID: <200EC2C0.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

I'd like to know a little more about this 'computer brat down the
street.' How do you know he is using SS? What is his output power if
he is using SS? When the SS guys across town get on the band, how much
power are they using? Are you sure they are "SS guys?"

What do these 'funny noises' sound like? I'm not trying to be
condescending, Brian, but I think you are mistaking packet radio for
SS.

73s de
Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1400] Shared SS
Author: hfsig@tapr.org at BALT.SMTP
Date: 8/1/96 9:37 AM

>But when the band gets loaded, and especially when the typical traffic
>consists of a lot of short, bursty transmissions, spread spectrum in
>general and frequency hopping in particular has the desirable effect
>of uniformly distributing the load over the band. The law of large
>numbers kicks in and the carrying capacity of the band increases
>thanks to the improved ability to dynamically share spectrum.

Narrowband users already dynamically share spectrum. They QSY to avoid QRM.
But if their frequencies were overlayed with SS, they would experience QRM
they couldn't evade.

Any attempt to mix SS and narrowband modes on HF will have to address the
following complaint from day one:

Before this thing you call spread spectrum came along, I could always have a
nice chat on 20-meter SSB. If QRM arose, we either convinced the other guy
to move or we moved ourselves. But now, whenever I listen on the band I
hear funny noises in the background. I can still hold a conversation but
it's not pleasant listening to this stuff. I tried QSYing to find a clear
spot but it didn't help--the damn stuff is everywhere. I missed a new
country last week when that computer brat down the street fired up on SS and
took out the whole band for hours. When the SS guys across town get on, I
can hear well enough to check into the Old Farts net, but I often miss my
turn when net control fades into their bleeps and bloops. Frankly, I don't
get on 20 much any more because it's just not worth the trouble. I want my
band back.

Any technical argument for SS that can't respond simply and directly to a
complaint like this will be immediately dismissed as BS. Shared SS won't
happen until hundreds of thousands of hams like the one imagined above are
convinced that SS will bring benefit instead of destroying their present ham
activities.

Brian Beezley, K6STI
k6sti@n2.net

7. Re: [HFSIG:1399] Re: Advantages of SS

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Thu, 1 Aug 1996 13:44:54 -0400
To: hfsig@tapr.org
Message-ID: <200ED5B0.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

Phil,

Could you PLEASE run for president of the ARRL?

73s de
Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1399] Re: Advantages of SS
Author: hfsig@tapr.org at BALT.SMTP
Date: 8/1/96 8:05 AM

Gwyn's comments are perhaps a bit blunt, but he's certainly on the
right track.

Another issue I've been meaning to comment on is the relative
efficiency of relay vs direct communications. The near-far problem
comes up mainly in the context of a station desiring to communicate to
a very distant point, typically requiring high power, while nearby SS
stations are communicating among themselves at relatively low power.

I believe Robert Glassey made an earlier comment along these lines
when he said that SS communications assume local cellular-style
communications. And he was quite right, if by "cellular style" he
meant heavy geographical reuse of spectrum, with relaying rather than
high power used to reach more distant stations. "Cellular style" does
NOT mean you're unable to talk DX, you just may not be able to do it
directly. (I routinely "work" hundreds of miles with my little 300mw
handheld 890 MHz unit...)

This strikes at another major aspect of ham radio: the
"self-sufficient ethic". Despite the literal meaning of the third
letter in "ARRL" the amateur culture strongly favors direct end-to-end
communications, especially in the HF bands. Repeaters are generally
not allowed on these bands, and even where they're allowed (e.g., on
10m) they are usually excluded from most contests as they're
considered to be "cheating".

Now some of this undoubtedly comes from the fact that historically,
repeaters consume more spectrum than simplex and are also limited
resources that are better reserved for other things, like mobile users
and emergencies. But there is also an underlying theme of
self-sufficiency that says a ham should have to do it all by himself to
gain credit. Big antennas and massive linear amplifiers command a lot
of respect, while resorting to a repeater is at least an admission of
inferiority, or even downright dishonest (if used in a competition).

Spread spectrum turns this culture on its head. Not only does the
average ham have to accept the obviously absurd notion that a wideband
signal can be more spectrally efficient than a narrowband signal, but
he is also being asked to believe that repeaters can be more
spectrally efficient than simplex! That is, a network of relatively
low power store-and-forward spread spectrum packet radios with
min-energy routing is far more spectrally efficient than direct high
power end-to-end operation.

Even the reliability aspect of not wanting to depend on a repeater
(sometimes cited in the context of emergency communications) is really
not an issue here. All you need are enough stations to back each other
up and you can have a network that, overall, is much *more* reliable
than DX simplex thanks to the shorter inter-node propagation paths.

All this has actually always been true even for narrowband modulation;
the near-far aspect of spread spectrum just makes it more glaringly
obvious. With cooperative relaying, it would still be possible to
work DX on HF. You just wouldn't necessarily do it
directly. Unfortunately, for a lot of hams this wouldn't be the
same.

Phil

8. Re: [HFSIG:1404] Re: Advantages of SS

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 1 Aug 1996 13:59:16 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608011859.NAA24314@www.kelly-afb.org>
Rendered body type: text/plain

Hear...hear...

>
>   Phil,
>
>   Could you PLEASE run for president of the ARRL?
>
>   73s de
>   Tony, KE4ATO
>
>
> _________________________ Reply Separator _____________________________
> Subject: [HFSIG:1399] Re: Advantages of SS
> Author: hfsig@tapr.org at BALT.SMTP
> Date: 8/1/96 8:05 AM
>
>
> Gwyn's comments are perhaps a bit blunt, but he's certainly on the
> right track.

9. Re: [HFSIG:1399] Re: Advantages of SS

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 01 Aug 1996 11:50:54 -0800
To: hfsig@tapr.org
Message-ID: <199608011902.MAA06926@PEAK.ORG>
Rendered body type: text/plain

Hi Folks,

I must say that I really enjoy the stimulating discussion about HF SS - extremely educational.

Personally, I'm not hearing convincing arguments either way, especially based on extreme situations or extrapolations.

Could someone please comment on the following:

1) If assumed that the center frequency of different DS SS operations are randomly chosen, what would that do to the near-far problem, i.e., if you use DS SS and experiencing interference from your neighbor DS SS operator, why not QSY out of harm's way of the offending signal envelope? This assumes that contacts are initially established on some known frequency. Am I right in assuming that if only two parties are involved in DS SS, that the PSD does have a maxima somewhere, even though it is a wideband, i.e, not quite white over the full bandwidth?

2) How many SS particpants does it take to start raising the noise floor to a point where it becomes objectionable? 2, 10, 100, 1000 ? If the number is 1000, would it be realistic, given a comparable narrow-band situation, to expect anyone to be able to conduct a QSO with that occupancy? Like a contest with 1000 RTTY operators calling simultaneously in the 14.090 - 14.065 20 meter segment.

As a lay person, and I admit I'm ignorant, the latter is the impression that I get from folks attempting to defend the narrow-band ideology. I'nt it time for someone to demonstrate a two-way digital QSO using SS on HF so we can all see for ourselves the magnitude of the real issues?

My two cents worth.

--Johan

10. Re: [HFSIG:1403] Re: Shared SS

From: Charles Brain <chbrain@dircon.co.uk>
Date: Thu, 01 Aug 1996 20:41:43 +0100
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19960801194143.0067b328@popmail.dircon.co.uk>
Rendered body type: text/plain

At 12:50 01/08/96 -0500, you wrote:
>   I'd like to know a little more about this 'computer brat down the
>   street.' How do you know he is using SS?

He isn't actually. In fact he is a member of Joe Public who has just turned his P.C on (The grey box with all the wires hanging out the back).

  :- }

- Charles

11. SS on HF bands

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Thu, 1 Aug 1996 13:31:39 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <199608012031.AA171931499@polecat.sr.hp.com>
Rendered body type: text/plain

Phil Karn <karn@qualcomm.com> said:

>So my main concern with the existing rules is that they keep us from
>using modulation and coding methods that conserve RF power. The result
>is a larger average amount of interference power to both narrowband
             ^^^^^^^
>and wideband users. ...

>Yes, it is possible to concoct worst cases where spread spectrum (or
>even a power-efficient wideband modulation scheme) could occasionally
>interfere with a nearby narrowband user. ...

>And this is ultimately why we should encourage spread spectrum on all
>of our bands, especially the most crowded ones.

Sigh. Let me try explaining this once again.

Spread spectrum works fine for something like a cellular telephone system
where the locations and power levels of all stations are well-controlled.
But on the amateur bands, stations are randomly distributed.

There is no need to "concoct worst case" scenarios: Most amateur
communication requires contacting a station that is much farther away
than other nearby stations.

If that nearby interfering station is using wide-band modulation, then the
interference potential is much greater, even though the *average* power
may be less. A decent communications receiver can handle narrowband
interference around 100 dB stronger than the noise level. Spreading the
interfering signal by, say, a factor of 100 only reduces the interference
by 20 dB. Narrowband has something like an 80 dB advantage over
broadband in this regard.

>I believe Robert Glassey made an earlier comment along these lines
>when he said that SS communications assume local cellular-style
>communications. And he was quite right, if by "cellular style" he
>meant heavy geographical reuse of spectrum, with relaying rather than
>high power used to reach more distant stations. "Cellular style" does
>NOT mean you're unable to talk DX, you just may not be able to do it
>directly. (I routinely "work" hundreds of miles with my little 300mw
>handheld 890 MHz unit...)

Cellular-style communications doesn't work on DX bands, because
nearby stations are typically very much weaker than distant stations.
(Because ground wave propagation is much lossier than sky wave.)

Even on the VHF/UHF bands, I have a hard time seeing how it would work.
If you are 100 yards from the nearest station, and 10 miles from the
next nearest (a not untypical situation), then you and your neighbor
will be able to relay between yourselves with no problem, but will have
trouble relaying to the next station in the chain over the +85 dB
interference from down the street. With narrow-band modulation, that
wouldn't be a problem.

>This strikes at another major aspect of ham radio: the
>"self-sufficient ethic". ...

(Much ad-hominum argument against narrow-band modulation omitted.)

>Spread spectrum turns this culture on its head. Not only does the
>average ham have to accept the obviously absurd notion that a wideband
>signal can be more spectrally efficient than a narrowband signal, ...

"Can be" but not "is" in this case. I guess I'm one of those ignorant
average hams...

AL N1AL

12. SS on HF

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Thu, 1 Aug 1996 15:06:04 -0800 (PDT)

To: hfsig@tapr.org
Message-ID: <199608012206.AA232007164@polecat.sr.hp.com>
Rendered body type: text/plain

forrerj@peak.org (Johan Forrer) said:

>2) How many SS particpants does it take to start raising the noise floor to
>a point where it becomes objectionable? 2, 10, 100, 1000 ?   ...

It only takes 1 if he is more than 20 dB or so louder than the signal you
are trying to receive. On HF, it is very rare to be working one of the
loudest signals on the band. Even for normal rag chewing, there are
typically a number of stations on the band that are much stronger than the
station you are talking to. When working a weak DX station, *most* of
the signals on the band are more than 20 dB louder. If even one of those
is running spread spectrum, you're out of luck.

If the SS station is running low power due to some dynamic power control
scheme, then that improves things somewhat. But even if that 20 dB number
becomes 30 or 40 dB, one SS station will still wipe out the band for a
large number of narrowband users. And by the way, other SS users will
have the same problem.

AL N1AL

13. Re: [HFSIG:1409] SS on HF

From: forrerj@peak.org (Johan Forrer.)
Date: Thu, 01 Aug 1996 16:59:06 -0800
To: hfsig@tapr.org
Message-ID: <199608020010.RAA06070@PEAK.ORG>
Rendered body type: text/plain

Hi Al,

>forrerj@peak.org (Johan Forrer) said:
>
>>2) How many SS particpants does it take to start raising the noise floor to
>>a point where it becomes objectionable? 2, 10, 100, 1000 ?   ...
>
>It only takes 1 if he is more than 20 dB or so louder than the signal you
>are trying to receive.
>On HF, it is very rare to be working one of the
>loudest signals on the band. Even for normal rag chewing, there are
>typically a number of stations on the band that are much stronger than the
>station you are talking to. When working a weak DX station, *most* of
>the signals on the band are more than 20 dB louder. If even one of those
>is running spread spectrum, you're out of luck.

To help me understand it a bit better: how much power (EIRP) would that
single SS station need to run to be 20 dB over your DX station?

Assume for argument sake that its a station in another state thats running
DS SS spread over 1 MHz bandwidth thats not skip to you; he's beaming in
your direction and that you receive the DX at S1 level?

--Johan

14. SS Stuff

From: Rick Booth <rick@itron-ca.com>
Date: Thu, 1 Aug 96 18:17:47
To: hfsig@tapr.org
Message-ID: <Chameleon.838949133.rick@rickb.itron-ca.com>
Rendered body type: text/plain

With regard to interference between SS systems and "conventional" users:

Consider a frequency hopper with the following properties. Using 2.4 KHz as the bandwidth of
the radio and assuming a baud rate of 300 (the most allowed by the FCC) and assuming 4 bits per
baud then in order to burst a bit sync preamble plus 10 bytes of data then the signal will have
to dwell on the frequency for about 84 milliseconds (80 bits plus 20 sync bits). For 40 meters
the band is only 300 KHz wide so the band is divided into about 125 slots to hop into. This
means every 10 seconds on the average the SS system will jump into a typical 2.4 KHz single
sideband "slot" and camp there for 84 milliseconds not counting filter thump from the 400
microsecond transient of a 2.4 KHz filter.

This might be annoying if the SS guy is down the street.

Rick
W6NZK

Rick Booth
'95 900SS SP
E-mail: rick@itron-ca.com
----

15. Re: [HFSIG:1411] SS Stuff

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 1 Aug 1996 23:42:11 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608020642.XAA28458@servo.qualcomm.com>
Rendered body type: text/plain

Rick seems to be describing a slow hopper, i.e., a system with
multiple data bits per hop. This produces a fairly long dwell time on
each channel. I was envisioning a "medium speed" hopper, with
something like one symbol per hop. That makes for a much faster hop
rate and a shorter dwell time on each channel.

As an aside, fast hopping is defined as multiple hops per modulation
symbol, and that implies a noncoherent combining loss because it's
difficult or impossible to maintain phase coherence across a
hop. (Even if the radios could do it, chances are you're hopping by
more than the coherence bandwidth of the channel.) Because of this
loss, fast hopping is generally recommended only against active
intelligent jamming, or when the coherence bandwidth of the channel is
unusually narrow, i.e., when the delay spread is long enough to
otherwise cause considerable ISI at the desired modulation symbol
rate.

Phil

16. Re: [HFSIG:1409] SS on HF

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 1 Aug 1996 23:59:10 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608020659.XAA28468@servo.qualcomm.com>
Rendered body type: text/plain

>It only takes 1 if he is more than 20 dB or so louder than the signal you
>are trying to receive. On HF, it is very rare to be working one of the

>becomes 30 or 40 dB, one SS station will still wipe out the band for a
>large number of narrowband users. And by the way, other SS users will
>have the same problem.

Again, this analysis seems to assume direct sequence. Frequency
hopping is quite different thanks to the bursty nature of the
interference.

Phil

17. Re: [HFSIG:1401] Re: Shared SS

From: Frode Weierud <frode@dxcern.cern.ch>
Date: Fri, 2 Aug 1996 10:09:07 +0200 (MET DST)
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <Pine.ULT.3.91.960802091805.15741A-100000@dxcern.cern.ch>
Rendered body type: text/plain

On Thu, 1 Aug 1996, Walt DuBose - K5YFW wrote:

>   Way back there in my AFRes days, we were running SSB on several
>   channels in the 8.4 - 8.9 MHz and in the 11.5 MHz range and the spooks
>   about 300 meters were running SS between 3 -15 MHz to the same
>   locations that we were. Propagation to the location we were both
>   talking to was fair on 8 MHz and poor on 11 Mhz with our equipment.
>   The spooks had very good communications. We were running 500 Watts
>   PEP and they were running less that a 100 watts IAW their specs.
>   Also, it didn't matter if they were on-the-air or not, it had no
>   affect on our communications and I don't recall seeing an
>   increase in channel noise...it darn sure didn't bother the marine
>   CW station on the channel ajacant to our main channel (frequency).
>   Also, there was another spook station located about 5 miles away
>   and we couldn't tell when either was transmitting unless we called
>   them on the fieldfone and asked. I might also mention that they
>   were running broadband antennas and we were running dipoles so our

> antennas were better.
>

As far as I know we already have SS transmissions on most if not all of our HF bands. Some years back I visited a military installation using SS on HF and I asked them if they blanked out the amateur bands and marine bands etc. The answer was NO. Their reply was that as we could not hear them they would not interfer and hence there was no reason not to include the ham bands in their frequency pools. I don't know what power they normally used, but they were equipped with 1KW and 10KW power amplifiers and the antennas were LPAs and disk-cones.

This is a one off experience, but I have the feeling that their attitude concerning SS and interference might be wide-spread and that many military/government organisations using SS will think the same way. Their main reason for using SS is the low probability of intercept and anti-jamming properties.

It would be interesting to hear if anybody else have some experience from military/government use of SS on HF, and specially if anybody is situated close to such an installation they could perhaps let us know if they have ever been able to detect the presence of SS signals.

73's Frode, F/LA2RL

Frode Weierud Phone : +41 22 7674794
CERN, SL, CH-1211 Geneva 23, Fax : +41 22 7679185
Switzerland E-mail : Frode.Weierud@cern.ch

18. SS and the Amateur culture

From: Robert.Glassey@nmp.nokia.com
Date: Fri, 2 Aug 96 12:51:50 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022934c2@MHS>
Rendered body type: text/plain

> I believe Robert Glassey made an earlier comment along these lines
> when he said that SS communications assume local cellular-style
> communications. And he was quite right, if by "cellular style" he
> meant heavy geographical reuse of spectrum, with relaying rather than

[snip] ... Yip, thats what I meant.

> This strikes at another major aspect of ham radio: the
> "self-sufficient ethic". Despite the literal meaning of the third
[snip]

> gain credit. Big antennas and massive linear amplifiers command a lot
> of respect, while resorting to a repeater is at least an admission of
> inferiority, or even downright dishonest (if used in a competition).
[snip]

> Spread spectrum turns this culture on its head. Not only does the
[snip]

> he is also being asked to believe that repeaters can be more
> spectrally efficient than simplex! That is, a network of relatively
[snip]

> up and you can have a network that, overall, is much *more* reliable
> than DX simplex thanks to the shorter inter-node propagation paths.
[snip]

> work DX on HF. You just wouldn't necessarily do it
> directly. Unfortunately, for a lot of hams this wouldn't be the
> same.

There's a few edited clips that I think sum up the angle you seem to be taking on this issue. I've already said my peice here, but I have just a few more comments.

You have quite rightly noted two things, 1. that spread spectrum culture is quite different from traditional amateur culture, and 2. that spread spectrum can make much more efficent *use* of RF resources.

One has got to ask, taking a wide view of amateur radio in general, how important ultimate efficency is, and why amateur culture is the way it is.

The valid arguments based on efficency are more in line with commercial objectives for the use of spectrum. They are more concerned with maximum number of users, level of service, and end user features. If amateur radio is operated according to such commercial objectives, it would make perfect sense for authorities to SELL the spectrum to commercial interests, since they will be able to provide the same type of service, suplimented with other telecoms, and may well be able to provide an even more efficent and reliable service, while making money for the authorities and the telecommunications industry.

There are already bands allocated for these services, and commercial interests are operating on them. Amateur radio has always *justfied*its* *existance*, not as an end user service provider (that all but regulated against), but as an experimental service for the self training of amateurs in the techniquies of radio communications. This is all encompassing, covering a wide variety of modes and applications and is for everyone from the beginner to the expert. One mode cannot take precedence at the expence of all others. Efficency at the expence of the aims and diversity of the service cannot be justified. As a cooperative additional mode, efficency would be welcomed, as was SSB. But SS has a long way to go before it could be accepted as such, if at all.

'You can have any colour you like, so long as its black.' - another expert in efficency and commercialism.

Rob

19. RE: [HFSIG:1410] Re: SS on HF

From: Jon Bloom <jbloom@connix.com>
Date: Fri, 2 Aug 1996 00:46:50 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB804F.370D2280@jbloom.connix.com>
Rendered body type: text/plain

Johan Forrer.[SMTP:forrerj@peak.org] wrote:
>To help me understand it a bit better: how much power (EIRP) would that
>single SS station need to run to be 20 dB over your DX station?
>
>Assume for argument sake that its a station in another state thats running
>DS SS spread over 1 MHz bandwidth thats not skip to you; he's beaming in
>your direction and that you receive the DX at S1 level?

Why would you assume he's in another state? I'd be more interested in the case where he's on the other side of town. Put it another way: that SS station may be in another state from me, but he's on the other side of town from *somebody.* With a decent receiver, I can handle a narrowband signal on the other side of town.

Over the past 20 years or so we hams have beat on the receiver manufacturers to make them design radios with 100+ dB of dynamic range. What that means is that we expect to be able to handle off-channel signals that are 100+ dB above the noise floor. The reason we want that kind of performance is because we *get* signal levels like that entering our receivers. Say there's a narrowband station nearby that I'm receiving at a level just within the 100-dB dynamic range of my receiver, so I'm not experiencing interference from his signal. If he now switches over to SS, he must reduce power and/or spread his signal to reduce the signal received in my receiver within the passband by 100 dB to be at my noise floor. (Of course, I'm willing to cut a few dB of slack.) Is that feasible? That's the question.

By the way, just what HF band do you plan to use for a 1-MHz-wide signal?

-- Jon

20. Re: [HFSIG:1407] Re: Shared SS

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Fri, 2 Aug 1996 09:09:30 -0400
To: hfsig@tapr.org
Message-ID: <201FFA80.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

Are you saying that his COMPUTER is causing the interference? I find
     it hard to believe he is causing that much interference 'down the
     street.'

     73s de
     Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1407] Re: Shared SS
Author: hfsig@tapr.org at BALT.SMTP
Date: 8/1/96 2:59 PM

At 12:50 01/08/96 -0500, you wrote:
> I'd like to know a little more about this 'computer brat down the
> street.' How do you know he is using SS?

He isn't actually. In fact he is a member of Joe Public who has just
turned his P.C on (The grey box with all the wires hanging out the back).

:- }

- Charles

21. Re: [HFSIG:1416] Re: SS on HF

From: forrerj@peak.org (Johan Forrer.)
Date: Fri, 02 Aug 1996 07:40:30 -0800
To: hfsig@tapr.org
Message-ID: <199608021452.HAA06591@PEAK.ORG>
Rendered body type: text/plain

Hi Jon,

>Why would you assume he's in another state? I'd be more interested in the
>case where he's on the other side of town. Put it another way: that SS
>station may be in another state from me, but he's on the other side of
>town from *somebody.* With a decent receiver, I can handle a narrowband
>signal on the other side of town.

I agree that local interference might be very strong. However, my experience
being an avid CW operator (I have a piece of wallpaper saying I won the CQWW
CW single band single op contest once upon a time), is the big signals from
elsewhere, not from local sources. So, thats what I was theorizing about a bit.

>
>Over the past 20 years or so we hams have beat on the receiver manufacturers
>to make them design radios with 100+ dB of dynamic range. What that means
>is that we expect to be able to handle off-channel signals that are 100+ dB
>above the noise floor. The reason we want that kind of performance is because
>we *get* signal levels like that entering our receivers. Say there's a
>narrowband station nearby that I'm receiving at a level just within the
>100-dB dynamic range of my receiver, so I'm not experiencing interference
>from his signal. If he now switches over to SS, he must reduce power and/or
>spread his signal to reduce the signal received in my receiver within the
>passband by 100 dB to be at my noise floor. (Of course, I'm willing to cut
>a few dB of slack.) Is that feasible? That's the question.

Yes, thats basically the gist of what I was after. How much power would it
take from a remote DS SS station to raise the noise floor to 20 dB over a
weak (say S1 level) DX signal. Just curious, because I suspect it would be
enormous (by amateur standards).

>
>By the way, just what HF band do you plan to use for a 1-MHz-wide signal?
>
>-- Jon
>
>

Just a theoretical question to see if I understand the nature of the SS PSD
implications. Perhaps you can give some insight; Would any degree of QSY
have an effect, i.e., how white is the spectrum really if there we have
light SS occupancy?

Thanks,

--Johan

22. Re: [HFSIG:1414] Re: Shared SS

From: Charles Hoffman <choffman@pelican.davlin.net>
Date: Fri, 02 Aug 1996 13:06:23 -0500
To: hfsig@tapr.org
Message-ID: <3202439F.3957@pelican.davlin.net>
Rendered body type: text/plain

Frode Weierud wrote:
>
> On Thu, 1 Aug 1996, Walt DuBose - K5YFW wrote:
>
> >   Way back there in my AFRes days, we were running SSB on several
> >   channels in the 8.4 - 8.9 MHz and in the 11.5 MHz range and the spooks
> >   about 300 meters were running SS between 3 -15 MHz to the same
> >   locations that we were.
> >   didn't matter if they were on-the-air or not, it had no
> >   affect on our communications and I don't recall seeing an
> >   increase in channel noise...it darn sure didn't bother the marine
> >   CW station on the channel ajacant to our main channel (frequency).
> >   Also, there was another spook station located about 5 miles away
> >   and we couldn't tell when either was transmitting unless we called
> >   them on the fieldfone and asked. I might also mention that they
> >   were running broadband antennas and we were running dipoles so our
> >   antennas were better.

        I have known Walt many years; his military comms experience is more
valid here than supposition of what a hypothetical computer guy down the
street will do or whatever hypothetical situation imagined. We need to
pay attention to Walt's experience reported here.
> >
>
> As far as I know we already have SS transmissions on most if not all of
> our HF bands. .. a military installation using SS
> on HF...asked them if blanked out the amateur bands and marine
> bands etc. The answer was NO... they were equipped with 1KW and 10KW power amplifiers ...antennas were
 LPAs

        Typical of HF SS mil/gov systems I am familiar with.

> Their main reason for using SS is the low probability of intercept and
> anti-jamming properties.

        Exactly the point!
>
> It would be interesting to hear if anybody else have some experience from
> military/government use of SS on HF, and specially if anybody is situated
> close to such an installation they could perhaps let us know if they have
> ever been able to detect the presence of SS signals.

        My amateur radio station is situated on a ranch in South Texas eight
miles directly off the backside of the recently installed "over the
horizion radar" (OHR) built by Raytheon. My best guess is that
collectively (multiple amps), feeding 13 phased LPA's run about 200,000
to the antennas (not erp). Frequency range, from observing the two
banks of LPAs appears to cover from 4 mHz to 30 mHz. The system
apparantly uses various modulation methods but the mode is SS, and it
does cover the complete spectrum including amateur radio frequencies.
        Considering the power, the nature of the LPA F/B characteristics and
the distance to the antenna, it is significant that I did not have any
harmful interference, including noticable receiver frontend sensitivity
reduction. During their testing phase I sometimes could tell when they
were on the air. Sometimes, and barely discernable. Later under normal
operations I never hear them.
        By comparison, from the same site they have two propogation measurement
devices. One is an oblique chirp sounder which squirts away from me and
the other is a F1F2 height sweep sounder using a NVIS dipole. The later
runs 10KW, and sweaps the spectrum of interest with what sounds like a
A1 carrier. It sweeps through my receiver passband in no more than
400ms but eats my lunch. The receiver is useless for three or four
seconds. The s-meter always hits the far peg with a "clink"!
        The LPA-backside SS signal is considerably greater rf field at my QTH
than the sounder signal but is unnoticed. I have heard it however; one
of the modes sounds like tinkling ice cubes in a pitcher of iced tea, as
its being poured. I do believe that is what killed the grass at the
ranch, not the drought.
        Hope this is of some interest. 73 Rik K5SBU
>
> 73's Frode, F/LA2RL
>
>   Frode Weierud   Phone : +41 22 7674794
>   CERN, SL, CH-1211 Geneva 23,   Fax   : +41 22 7679185
>   Switzerland   E-mail : Frode.Weierud@cern.ch

23. SS on HF

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Fri, 2 Aug 1996 13:00:07 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <199608022000.AA220946008@polecat.sr.hp.com>
Rendered body type: text/plain

forrerj@peak.org (Johan Forrer.) said:

>>>2) How many SS particpants does it take to start raising the noise floor to
>>>a point where it becomes objectionable? 2, 10, 100, 1000 ? ...
>>
>>It only takes 1 if he is more than 20 dB or so louder than the signal you
>>are trying to receive.
>
>To help me understand it a bit better: how much power (EIRP) would that
>single SS station need to run to be 20 dB over your DX station?
>
>Assume for argument sake that its a station in another state thats running
>DS SS spread over 1 MHz bandwidth thats not skip to you; he's beaming in
>your direction and that you receive the DX at S1 level?

The propagation loss varies widely with different paths. At any given time,
signals on a DX band typically vary from the noise level (less than a tenth
of a microvolt, around -130 dBm) up to the millivolt range, -30 dBm or so.
(This assumes skywave propagation. It can be worse if there is another ham
station in your neighborhood.) A good communications receiver's blocking
level might be around 0 dBm, so it can handle typical band conditions with
no problems. But if one of those -30 dBm signals has spread its signal
over a 300 kHz band, it will be putting -50 dBm interference into each
3 kHz narrow-band channel.

You would have to reduce the power by something on the order of 80 dB in
order not to cause interference. 80 dB below a legal-limit 1500W signal is
15 microwatts, much too low for reasonable communications capability on HF.
You can argue with my numbers, but you would have to find a very large
error in them in order to invalidate my argument.

If the spread spectrum signal is a frequency-hopping type, then it will
be putting a full-strength (non-spread) signal into each channel at some
repetitive rate. In order to cause no interference at all, the power must
be reduced even 20 dB more than for the example above. True, frequency
hopping does reduce the percentage of time that the interference occurs,
but that percentage increases the more SS stations there are operating
on the band at any given time.

AL N1AL

24. Re: [HFSIG:1400] Shared SS

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Fri, 2 Aug 1996 15:46:39 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608022246.PAA05002@unix.ka9q.ampr.org>
Rendered body type: text/plain

>Narrowband users already dynamically share spectrum. They QSY to avoid QRM.

That's not dynamic enough. You're still dedicating a chunk of spectrum
to a particular QSO for as long as it lasts. It goes to waste whenever
neither party is talking, and even during speech pauses during a transmission.

The kind of dynamic sharing I'm talking about exploits the
peak/average ratios of voice conversation, not to mention the
enormously larger ratios present in most computer commnications that
is the rationale for packet switching. It puts all the users and all
the spectrum in one big pool to exploit the law of large numbers.

Qualcomm CDMA does exactly this with its variable rate vocoder. When
you talk, the data rate kicks up to 9.6 kb/s (or 14.4 kb/s in newer
systems). When you stop talking, it drops back to 1.2 kb/s. The average
data rate is something like 40% of the peak in a typical conversation.

A typical CDMA sector handles 25-35 calls in each 1.25 MHz RF channel,
and that's enough to gain considerable benefit from pooling. Only if
everyone talks at exactly the same moment do you hit the limit, and
the probability of that goes down as you make the pool larger.

Again, I won't give you a guarantee that SS will *never* cause
interference. I can't, though I can certainly improve the odds with
power control, efficient coding and adaptive hopping patterns. But
it's just unrealistic to hold SS to such a high standard when

narrowband operations are not. QRM has always been a way of life on
the shared HF bands, and the best we can do is to promote methods that
reduce it on average.

Phil

25. Re: [HFSIG:1420] SS on HF

From: forrerj@peak.org (Johan Forrer.)
Date: Fri, 02 Aug 1996 16:26:23 -0800
To: hfsig@tapr.org
Message-ID: <199608022338.QAA01803@PEAK.ORG>
Rendered body type: text/plain

Alan,

> A good communications receiver's blocking
>level might be around 0 dBm, so it can handle typical band conditions with
>no problems. But if one of those -30 dBm signals has spread its signal
>over a 300 kHz band, it will be putting -50 dBm interference into each
>3 kHz narrow-band channel.
>
>You would have to reduce the power by something on the order of 80 dB in
>order not to cause interference. 80 dB below a legal-limit 1500W signal is
>15 microwatts, much too low for reasonable communications capability on HF.
>You can argue with my numbers, but you would have to find a very large
>error in them in order to invalidate my argument.
>

This is exactly what I was trying to grasp. Thanks for the explanation.

--Johan

26. HF SS Demonstration

From: k6sti@n2.net (Brian Beezley)
Date: Fri, 2 Aug 1996 23:26:18 -0700
To: hfsig@tapr.org
Message-ID: <199608030626.XAA06402@ravel.n2.net>
Rendered body type: text/plain

At this point it seems like everyone has pretty much spoken his piece
regarding the feasibility of mixing spread spectrum and existing HF
narrowband modes.

I'd like to take the discussion one step further by proposing a simulated HF
SS demonstration. The demonstration will give anyone with an HF receiver a
chance to hear what frequency-hopped signals sound like and to judge their
effect on narrowband users.

Here's what I propose: I'll write PC software that hops an HF transceiver
around in the 20-meter band. We'll run the transceiver in USB mode and
drive it with a controlled audio waveform generated by a sound card. We'll
pick a contest-free Saturday and for a 15-minute period, a dozen or more
stations across the country simultaneously will transmit simulated SS.
Interested parties can listen to the band during this demonstration and make
their own judgment about the level of interference generated. You can
monitor conversations in progress and listen for comments. You can record
what you hear. If you're near one the simulated SS stations, you can judge
the level of near-transmitter QRM; otherwise, you'll have to be content to
listen to the skywave signals.

I'll leave it to the SS proponents to specify a power level and devise a
waveform and hopping sequence. I'd suggest a maximum of 50 watts RMS output
power to avoid overheating barefoot exciters. I'd suggest using a noiselike
waveform rather than one with spectral lines to make it audibly less
intrusive. For this demo I'm not volunteering to program an adaptive hop
algorithm--to do it right would take more time and effort than I'm willing
to expend on this experiment.

To simplify things, I'll write the code to control the frequency of a single
brand of transceiver. Unfortunately, I don't own a transceiver that can be
computer controlled, but I do have access to several locally for testing.

There are a couple of nontechnical issues involved in a test like this. The
first is the propriety of intentionally emitting power on the frequency of a
QSO in progress without permission or warning. The second concerns the
legality of such a test. I expect these issues to be addressed in
discussion that follows by those concerned. Personally, I don't have a
problem with either as long as the test is brief. Presumably, none of the

stations volunteering to transmit SS will either.

I don't propose to publicize this demonstration ahead of time except to
announce the time and date on several Internet reflectors a day in advance.

If the SS proponents aren't interested in this demonstration or if I can't
scrounge up enough volunteers, I'll drop the idea. Otherwise, who's game?

Brian Beezley, K6STI
k6sti@n2.net

27. Re: [HFSIG:1401] Re: Shared SS

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sun, 4 Aug 1996 14:49:40 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608042149.OAA00779@unix.ka9q.ampr.org>
Rendered body type: text/plain

Way back there in my AFRes days, we were running SSB on several
        channels in the 8.4 - 8.9 MHz and in the 11.5 MHz range and the spooks
        about 300 meters were running SS between 3 -15 MHz to the same
        locations that we were. Propagation to the location we were both

To be perfectly fair to the "other side", I have to point out that the
process gain (spreading range) here was much larger than a single ham
band. If frequency hopping were used, I'd expect them to be virtually
undetectable except on a wideband spectrum analyzer close to their
transmitter.

Phil

28. Re: [HFSIG:1424] Re: Shared SS

From: Charles Hoffman <choffman@pelican.davlin.net>
Date: Sun, 04 Aug 1996 18:18:35 -0500
To: hfsig@tapr.org
Message-ID: <32052FCB.4A9A@pelican.davlin.net>
Rendered body type: text/plain

Phil Karn wrote:
>
>   Way back there in my AFRes days, we were running SSB on several
>   channels in the 8.4 - 8.9 MHz and in the 11.5 MHz range and the spooks
>   about 300 meters were running SS between 3 -15 MHz to the same
>   locations that we were. Propagation to the location we were both
>
> To be perfectly fair to the "other side", I have to point out that the
> process gain (spreading range) here was much larger than a single ham
> band. If frequency hopping were used, I'd expect them to be virtually
> undetectable except on a wideband spectrum analyzer close to their
> transmitter.
>
> Phil

        Interesting, but who's watching? Just be invisible like everyone else
on SS! Ho ho. How many amateur radio stations are on the ham bands on
SS? How many non amateur stations are on the ham bands on SS? Does it
matter? Who is minding the store?

        73 Rik K5SBU

29. Re: [HFSIG:1424] Re: Shared SS

From: fperkins@onramp.net
Date: Sun, 4 Aug 1996 18:33:16 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608042333.SAA13962@mailhost.onramp.net>
Rendered body type: text/plain

Hi Phil and the HF Gang,

I looked around a bit this weekend trying to find a basic discussion
on the channel capacity of "random" packet transmissions. I am trying
to gain some insight into the "18% barrier".

Could you suggest a reference?

73 Frank WB5IPM

30. Re: [HFSIG:1402] Wide band modes

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sun, 4 Aug 1996 17:34:40 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608050034.RAA00887@unix.ka9q.ampr.org>
Rendered body type: text/plain

Robert, thanks for your long and thoughtful note. There was so much in there, I just finished digesting it all.

>It is indeed. I have no arguments here. I've long suspected as much, >although I've never had the numbers that show it. Although even very >simple ECC can make back most of this 35-40db difference. ie and AMTOR

Quite true. To paraphrase your local real estate agent, three things are most important in dealing with Rayleigh fading: diversity, diversity and diversity. Even simple repetition coding (the simplest form of ECC) is a significant benefit. But properly constructed codes use the redundancy more efficiently, and work better as a result.

>I guess without any extra power, the signal would be above the threshold >50% of the time. Thus we have a 50% BER. A 1/2 rate ECC would ideally >deal with this, although I guess a bit more redundancy would be usefull.

A 50% BER corresponds to zero capacity, which is the worst any communication system can do. (If the BER were greater than 50%, you could simply invert it to make the BER less than 50%, and you'd have some non-zero capacity).

Since the capacity of a BSC (binary symmetric channel) with a 50% BER is zero, no code can possibly fix this. However, a BEC (binary erasure channel, i.e., a BSC supplemented with a third "erasure" output) is a different story. The capacity of a BEC, assuming no "hard" errors, is as you expect very close to 1 minus the erasure rate. A rate 1/2 ECC would work pretty well here assuming proper attention is given to interleaving.

>In the light of this 15dB fade margin, I wonder how much effect soft >decoding would really have. Would erasure decoding be just as good? >What I'm geting at is PACTOR II's K=9 or is it 11 contraint length. >Would k=5 be just as good? Or could we just go for plain Reed Soloman >coding? This would make PC implementation of an HFSIG protocol more >realistic.

This is a very interesting question. You can think of erasure decoding as a form of "softened" decision decoding where the received symbols can take on three levels (zero, one and "don't know"), and it's certainly true that a little bit of "softness" goes a long way. It does make Reed-Solomon decoders look more interesting since they can handle erasures fairly easily but not full-blown soft decisions.

But I see nothing wrong with Viterbi decoding per se, and there is probably still some gain to be had from soft decision decoding. I think I have some references that can quantify this. Sure, Viterbi is slower than Reed-Solomon but with all the cycles we have these days it's just not a major factor at the data rates we're likely to use. (My K=7 decoder runs at 155 kb/s on a P90).

>But getting back to your spreading argument, I've just suggested >trippling the bandwidth to overcome fading, or have I?

Exactly!! FEC inherently "spreads" bandwidth, and that makes it as much a "spread spectrum" scheme as frequency hopping or direct sequence chipping. It certainly does cause you to run into the FCC emission bandwidth limits, which are the basic problem here.

>I recall Ungerboch (I can never spell that) said that the solution was >to go for multi levels, even if you cannot actually resolve each one

I'm not surprised Ungerboch said this. He invented trellis coding, and it's been extremely successful on bandwidth limited channels (e.g., telephone modems). But beyond the inappropriate use of the bandwidth-limited model for the power-limited radio channel, fading introduces some serious practical problems. If the symbols have unequal energy, the demodulator thresholds would depend on signal amplitude, and this changes rapidly and unpredictably with fading. This is in fact one of the most serious drawbacks of on-off-keyed CW. Ever tried to copy it on a channel that's fading at about the same rate as a CW symbol element?

In my opinion, all these higher-order QAM schemes in Clover et al are
going down the wrong path. They certainly help account for their
extremely high Eb/N0 requirements when operating at high data
rates. It would be much better to let them relax their bandwidth
constraints and just scale up their more power-efficient schemes like
BPSK to wider channels.

>Not inevitible, but I can still see that a wider band signal will have
>faster fades that are more easilly dealt with. I think the wider
>bandwidth helps when you consider interference from other users. Still a
>narrow a bandwidth allows more users in the same wide bandwidth without
>interference. 6 of one, half dozon of another as they say. I dont think
>either can make full use of the bandwidth.

It's not that the fading is faster in the wideband case, it's just
another way to get more diversity.

You can get diversity in different ways: time, frequency, space,
polarization, etc. All that matters is that the diversity is over
sufficient distance to get some independence in the fading. That is,
you need to diversify the effect of any given user data bit over
considerably more than one coherence time, one coherence bandwidth,
etc. It's sufficient to diversify over just one dimension, but you'll
need more of it than if you use several. For example, real-time voice
may have some serious delay constraints that limit the use of time
diversity, so frequency diversity is perhaps even more useful than it
would be in a non-real-time packet data system where you could afford
to diversify over many seconds of time.

I disagree strongly with the "6 of one, half dozen of the other"
characterization. It is very clearly the case that what you spend in
bandwidth is more than bought back in extra interference resistance.
This is most clearly the case when the extra bandwidth allows a decrease
in Eb/N0 through coding, but there are other practical benefits from
spreading even when the AWGN Eb/N0 is unaffected (as it is with direct
sequence or frequency hopping): the increased "trunking" or "pooling"
efficiencies of a large group of users dynamically sharing a large pool
of spectrum, and the benefits of frequency diversity that I've already
mentioned.

>Hmm, this is interesting, expecially the 5dB increase over the BPSK
>case. I wonder what it would be over the 64-QAM case.

The literature I have doesn't give any results for QAM since it's
considered in appropriate for the fading channel. Even coherent BPSK
is inappropriate because of the difficulty in tracking carrier phase
(differentially detected BPSK is recommended instead.)

>How resistant would this M-ARY scheme be to other signals? I wonder if

Depends on the nature of the QRM. Continuous unmodulated carriers
could be handled rather easily with adaptive notch filters. But strong
intermittent QRM would be harder to deal with. That's why you really
want to combine this with frequency hopping; by using only a small
subset of all the available frequencies at any given time, you reduce
the chances of landing on strong QRM to where coding can easily handle it.

>I'm actually fairly well convinced that with a properly designed digital
>radio, with a wide bandwidth and demod controlled AGC could overcome the
>near far problem quite well, especially if spread enough so that local
>impulses cause a BER of < 10%. I even like the idea!

Great!

>The concern is about sharing the band with other users. I dont think

I agree, and actually that's why I would support dividing up the band
into "DX" and "regional" subbands to mitigate the near-far problem.
Each subband would still allow wideband modulation within it, but only
for the right purpose (i.e., DX or local). I think this would go a
long way towards solving the problem, don't you?

Phil

31. Re: [HFSIG:1406] Re: Advantages of SS

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sun, 4 Aug 1996 22:00:45 -0700 (PDT)

To: hfsig@tapr.org
Message-ID: <199608050500.WAA01008@unix.ka9q.ampr.org>
Rendered body type: text/plain

>1) If assumed that the center frequency of different DS SS operations are
>randomly chosen, what would that do to the near-far problem, i.e., if you
>use DS SS and experiencing interference from your neighbor DS SS operator,
>why not QSY out of harm's way of the offending signal envelope? This assumes
>that contacts are initially established on some known frequency. Am I right
>in assuming that if only two parties are involved in DS SS, that the PSD
>does have a maxima somewhere, even though it is a wideband, i.e, not quite
>white over the full bandwidth?

Johan,

You assume that there's little if any filtering after the spreading
operation. Most real systems I know do in fact filter, both to
flatten the in-band spectrum and to knock off the sin(x)/x
sidelobes. Qualcomm CDMA uses a particularly sharp FIR filter because
of cellular carrier requirements. To get the sharp skirts we let the
passband ripple somewhat. The resulting display on a spectrum analyzer
closely resembles the top of Bart Simpson's head, so around here we
often call each composite CDMA waveform a "Bart's head".

>2) How many SS particpants does it take to start raising the noise floor to
>a point where it becomes objectionable? 2, 10, 100, 1000 ? If the number is
>1000, would it be realistic, given a comparable narrow-band situation, to
>expect anyone to be able to conduct a QSO with that occupancy? Like a
>contest with 1000 RTTY operators calling simultaneously in the 14.090 -
>14.065 20 meter segment.

This depends on the process gain, required Eb/N0s and relative signal
strengths. The math is actually pretty straightforward.

Consider a DS/SS system with coding and spreading such that an Eb/N0
of 6 dB is required and the ratio of RF bandwidth to user data rate is
128:1. (These are the approximate parameters of Qualcomm CDMA).

The process gain is 10*log10(128) ~= 21 dB. For a required Eb/N0 of 6 dB,
the overall SNR required is 6 - 21 = -15 dB. This figure is sometimes
referred to as Ec/I0, the ratio of the energy per chip to the interference
spectral density.

If everyone's power is matched at the receiver, you can tolerate as
many as 31.6 other users before their aggregate interference brings
the Eb/N0 down to 6 dB. Note this is less than the 128 users you could
theoretically support by just dividing up the total bandwidth into 128
equal size pieces. But that assumes no thermal noise, i.e., that it
would be easy to operate at the higher Eb/N0 that's required without
bandwidth-expanding coding, especially on a fading channel. It also
ignores any savings from better dynamic sharing, and it ignores the
benefits from being able to geographically reuse frequencies more
densely because of the far greater resistance to co-channel
interference.

Phil

32. Re: [HFSIG:1408] SS on HF bands

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sun, 4 Aug 1996 22:26:25 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608050526.WAA01021@unix.ka9q.ampr.org>
Rendered body type: text/plain

>Spread spectrum works fine for something like a cellular telephone system
>where the locations and power levels of all stations are well-controlled.
>But on the amateur bands, stations are randomly distributed.

Most cellular users are randomly distributed too. But they do generally
talk only to the nearest base station.

A somewhat better model is the DARPA SURAN (Survivable Radio Network)
project of the 1970s and 1980s. It assumed randomly distributed packet
radios operating in a simplex fashion. It used spread spectrum.

>If that nearby interfering station is using wide-band modulation, then the
>interference potential is much greater, even though the *average* power
>may be less. A decent communications receiver can handle narrowband
>interference around 100 dB stronger than the noise level. Spreading the
>interfering signal by, say, a factor of 100 only reduces the interference

>by 20 dB. Narrowband has something like an 80 dB advantage over
>broadband in this regard.

This is true as far as it goes, but 100 dB adjacent channel rejection
seems a little generous. 60 dB is a more reasonable estimate, I think.
Whatever.

>Cellular-style communications doesn't work on DX bands, because
>nearby stations are typically very much weaker than distant stations.
>(Because ground wave propagation is much lossier than sky wave.)

Ah, but wouldn't this tend to mitigate the near-far problem?

>(Much ad-hominum argument against narrow-band modulation omitted.)

I don't see why you have to label this "ad-hominum". I think it's a
fair characterization of the "ham ethic" as I see it.

I guess it boils down to what you think ham radio is all
about. Reliable communications? That description hardly fits direct
long-haul HF. Easily accessible to those without large lots, liberal
housing covenants and money for big antennas and power amplifiers? I
think not. The popularity of VHF/FM and repeater networks shows there
are a lot of hams who are interested primarily in reliable local
communications with small stations, and who aren't purists when it
comes to using relays. Spread spectrum promises to provide the same
capabilities over wider areas on HF.

I confess that perhaps my view of ham radio is somewhat different than
yours. About the only activity that has a hope of justifying our bands
for very long is technical self-training, experimentation and
advancement. That means keeping up with the state of the art in the
non-amateur world, and spread spectrum is certainly one technology
where we trail the commercial and military world. While I don't really
have anything against DXing and contesting, I really don't see how
they help justify ham radio's existence to the outside world; they're
really little more than technological stamp-collecting.

I don't want to do away with traditional hamming, but I don't want it
to choke experimentation either. While I can't guarantee that there
won't ever be any SS interference to narrowband modes, I think it can
be minimized by careful system design and operational coordination.
And whatever risks there are will be worth the benefits. After all,
ours is not a critical safety-of-life service like aviation; we're
supposed to be experimenters, and experiments can fail. That's part of
the process.

Phil

33. Re: [HFSIG:1415] SS and the Amateur culture

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sun, 4 Aug 1996 23:36:58 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608050636.XAA01066@unix.ka9q.ampr.org>
Rendered body type: text/plain

>One has got to ask, taking a wide view of amateur radio in general, how
>important ultimate efficency is, and why amateur culture is the way it
>is.

A very good question.

>There are already bands allocated for these services, and commercial

>interests are operating on them. Amateur radio has always *justfied*its*
>*existance*, not as an end user service provider (that all but regulated
>against), but as an experimental service for the self training of
               ^^^^^^^^^^^^^^^^^^^^^^^^^^^^

BINGO!!

How much serious, state-of-the-art experimentation is going on right now
in amateur radio? You mention single sideband, but that was a long time
ago. As the saying goes, what has ham radio done for the world lately?

Indeed, ham radio doesn't exist in a vacuum. It is not just a hobby.
It is given spectrum that could otherwise be sold for big bucks
because it supposedly gives something back to society as a whole. Now
that emergency communications and international good will have largely

been taken over by cellular telephones and the Internet, respectively, what's left? Technical experimentation and self-training! How much of that really goes on in ham radio these days? Not much. Even worse, the rules are set up to discourage it. That's what I want to change.

Phil

34. Re: [HFSIG:1426] Re: Shared SS

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 5 Aug 1996 00:36:30 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608050736.AAA05953@servo.qualcomm.com>
Rendered body type: text/plain

>on the channel capacity of "random" packet transmissions. I am trying
>to gain some insight into the "18% barrier".

That's the original Aloha result. Keep in mind the conditions: a single narrowband channel; two or more transmissions always result in a collision, i.e., there is no capture effect; there is no channel sensing, so stations transmit whenever they want.

Spread spectrum has entirely different results because collisions are "soft". That is, you can have many stations transmitting at once. At some point the background interference will be too great to get through, but that happens at a load much greater than 2 active transmitters. And again the law of large numbers kicks in, so you can get much closer to the carrying capacity of the channel than you could on a single narrowband (no capture) channel.

Phil

35. Re: [HFSIG:1402] Wide band modes

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Mon, 5 Aug 1996 00:59:48 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608050759.AAA01117@unix.ka9q.ampr.org>
Rendered body type: text/plain

>How resistant would this M-ARY scheme be to other signals? I wonder if
>different convolution codes could be used to provide further resistance
>between signals (getting very spread spectrum like now), with some
>sequential soft decoding being used to distinguish between the different
>stations, rather than just using a majority vote detector. Could this
>provide enough resistance to other stations on the same channel? It may
>be possible to decode 100% a weaker station while a much stronger
>station, or two use the same channel. MARY FSK seems to have quite an
>advantage here.

I meant to comment on the latter part of this paragraph.

You are exactly right -- different convolutional codes can indeed be used to differentiate between signals; it's an entirely viable form of CDMA. If the code rate is low enough, then this is virtually identical to coded direct sequence spread spectrum! In fact, Andrew Viterbi invented such a scheme a few years ago and published a paper about it.

I'm not sure it is of real benefit, though, since you can do almost as well by using a common high rate code (e.g., rate 1/3 or 1/2) and then chipping the encoded symbols with different PN codes in a more conventional direct sequence system.

Although the lower rate code of Viterbi's scheme would give somewhat greater coding gain, most of the gain in a code comes in the first factor of 1/2-1/4 or so. Ultra-low code rates with distinct polynomials for each encoded symbol doesn't gain enough to be worthwhile yet from a complexity standpoint over simply repeating the symbols from a relatively high rate coder -- and that's what the PN chipping effectively does.

Again, the name of the game on a fading channel is *diversity*. Coding and interleaving gives you diversity in the time domain, assuming the interleaving span is much greater than the coherence time. It can also give you some diversity in the frequency domain if the code rate is low enough to increase the signal bandwidth significantly beyond the coherence bandwidth. But just as simple repetition is a viable (if not ideal) form of diversity in the time domain, so it is in the frequency domain -- and that's what direct sequence chipping or frequency hopping both do.

BTW, the original wideband system that John Costas analyzed in his
1959 paper "Poisson, Shannon and the Radio Amateur" was essentially
simple "frequency repetition". That is, he divided up his RF energy
across N identical copies in frequency of the analog waveform and
recombined them on the receiving end. Even with this relatively weak
form of coding (a scheme he described as "intuitively ridiculous"), he
found noticeable improvements in band capacity.

Phil

36. Military SS systems

From: Robert.Glassey@nmp.nokia.com
Date: Mon, 5 Aug 96 11:52:51 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022b01a5@MHS>
Rendered body type: text/plain

On Thu, 1 Aug 1996, Walt DuBose - K5YFW wrote:

> Way back there in my AFRes days, we were running SSB on several
> channels in the 8.4 - 8.9 MHz and in the 11.5 MHz range and the
> spooks about 300 meters were running SS between 3 -15 MHz to the same
> locations that we were. didn't matter if they were on-the-air or not,
> it had no affect on our communications and I don't recall seeing an
> increase in channel noise...it darn sure didn't bother the marine CW
> station on the channel ajacant to our main channel (frequency). Also,
> there was another spook station located about 5 miles away and we
> couldn't tell when either was transmitting unless we called them on
> the fieldfone and asked. I might also mention that they were running
> broadband antennas and we were running dipoles so our antennas were
> better.

Rik K5SBU wrote:

> My amateur radio station is situated on a ranch in South Texas eight
> miles directly off the backside of the recently installed "over the
> horizion radar" (OHR) built by Raytheon. My best guess is that
> collectively (multiple amps), feeding 13 phased LPA's run about
> 200,000 to the antennas (not erp). Frequency range, from observing
> the two banks of LPAs appears to cover from 4 mHz to 30 mHz. The
> system apparantly uses various modulation methods but the mode is SS,
> and it does cover the complete spectrum including amateur radio
> frequencies.
> Considering the power, the nature of the LPA F/B characteristics and
> the distance to the antenna, it is significant that I did not have any
> harmful interference, including noticable receiver frontend
> sensitivity reduction.

These are very interesting results. Is you station behind the LPAs? Do
they ever point in your direction? Have you attempted (out of interest)
to measure interference in front the the LPA's say at a mile or two? Do
the LPAs point away from built up areas?

Over the weekend I did a series of tests on 18MHz with low power pulsed
transmissions to determine their range and interference potential. I'll
post a report soon, however I was able to hear 10mW 2ms carrier bursts
at a distance of 1 mile, and 1W 2ms bursts at 3 miles, using a handheld
scanning receiver with a 70cm long whip antenna!

So how do the military do it? Are they really using kilowatts of power?
How short are their bursts?

It would be very interesting to get a wide band receiver (4-30MHz) with
a fast envelope detector, and very fast peak hold and bring it close to
the SS instalation in Texas. Surely this would make the TX power
detectable, allowing comparisons with other TX sources, and giving an
idea of how the signal is spread.

I expect the military have spent millions on reducing interference (Low
probability of detection) and they use extreemly wide bandwidths. A
radar may well have quite different characteristics compared to a
communications system.

I expect most of their techniques are classified, but there must be
something we can learn from them. From my tests a simple hopping 2ms
burst is unacceptable. How do they do it??

My tests suggest that a DSSS system with 10 microwatts power in 2.4kHz

(100mW in 24MHz) would be detectable at 3 miles. A 10mW carrier 2.4kHz
SNR is about 0dB (still audible above the noise) at 3 miles using a 70cm
long unmatched whip antenna - shown to be at least 30dB less sensitive
than a reasonable amateur station in the presence of ambient noise.

Cheers,

Rob, G0VTQ

37. [HFSIG:1423] HF SS Demonstration

From: Robert.Glassey@nmp.nokia.com
Date: Mon, 5 Aug 96 12:07:09 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022b01b2@MHS>
Rendered body type: text/plain

Hi Brian.

> I'd like to take the discussion one step further by proposing a
> simulated HF SS demonstration.

Aha! exactly what I was thinking. I did just this last weekend. I'll
post a report soon.

> Here's what I propose: I'll write PC software that hops an HF
> transceiver around in the 20-meter band. We'll run the transceiver in
> USB mode and drive it with a controlled audio waveform generated by a
> sound card.

We think alike! This is almost exactly how I did it! except I only used
one SSB channel and transmitted no output when the frequency should have
been in some other channel.

> We'll pick a contest-free Saturday and for a 15-minute
> period, a dozen or more stations across the country simultaneously
> will transmit simulated SS.

.. and I only had one TX and measured interference at a range of 1-5
miles with a hand held scanner.

> I'll leave it to the SS proponents to specify a power level and devise
> a waveform and hopping sequence. I'd suggest a maximum of 50 watts
> RMS output power to avoid overheating barefoot exciters. I'd suggest
> using a noiselike waveform rather than one with spectral lines to make
> it audibly less intrusive.

Tried this, but this form of partical spread does not appear to make the
sound any more acceptible, or more dificult to hear. My tests were 10mW
to 1 Watt, and resulted in significant interference.

Rob

38. Re: [HFSIG:1434] HF SS Demonstration

From: Mike Murphree <n4cnw@pig.net>
Date: Mon, 05 Aug 1996 09:01:19 -0400
To: hfsig@tapr.org
Message-ID: <3205F09F.33C0@pig.net>
Rendered body type: text/plain

Robert.Glassey@nmp.nokia.com wrote:

> > Here's what I propose: I'll write PC software that hops an HF
> > transceiver around in the 20-meter band. We'll run the transceiver in
> > USB mode and drive it with a controlled audio waveform generated by a
> > sound card.
>
> We think alike! This is almost exactly how I did it! except I only used
> one SSB channel and transmitted no output when the frequency should have
> been in some other channel.
>
> . and I only had one TX and measured interference at a range of 1-5
> miles with a hand held scanner.
> Tried this, but this form of partical spread does not appear to make the
> sound any more acceptible, or more dificult to hear. My tests were 10mW
> to 1 Watt, and resulted in significant interference.

This sounds pretty slow.. What was the dwell time on each frequency?
What kind of transceiver was used? DDS?

What modulation was used?

Mike

39. Re: [HFSIG:1434] HF SS Demonstration

From: k6sti@n2.net (Brian Beezley)
Date: Mon, 5 Aug 1996 06:23:28 -0700
To: hfsig@tapr.org
Message-ID: <199608051323.GAA04174@ravel.n2.net>
Rendered body type: text/plain

>> I'd suggest using a noiselike waveform rather than one with spectral lines >> to make it audibly less intrusive.
>
>Tried this, but this form of partical spread does not appear to make the
>sound any more acceptible, or more dificult to hear. My tests were 10mW
>to 1 Watt, and resulted in significant interference.
>
>Rob

Odd. I find stations tuning up--even weak heterodynes--a lot more annoying than SSB splatter or the occasional solar noise burst. The San Diego county noise ordinance lowers the one-hour average SPL noise limit by 5 dB if the noise contains a steady, audible tone "such as a whine, screech, or hum."

Brian Beezley, K6STI
k6sti@n2.net

40. Re: [HFSIG:1433] Military SS systems

From: Charles Hoffman <choffman@pelican.davlin.net>
Date: Mon, 05 Aug 1996 09:17:16 -0500
To: hfsig@tapr.org
Message-ID: <3206026C.111F@pelican.davlin.net>
Rendered body type: text/plain

Robert.Glassey@nmp.nokia.com wrote:
>

> Rik K5SBU wrote:
>
> > My amateur radio station is situated on a ranch in South Texas eight
> > miles directly off the backside of the recently installed "over the
> > horizion radar" (OHR) built by Raytheon.

> These are very interesting results. Is you station behind the LPAs?

Yes, eight miles.

> Do they ever point in your direction?

No, the array is composed of a total of 26 seperate LPA's in vertical configuration, each suspended between towers, and parallel with the next one. The entire antenna array looks from a distance to be about 50 meters by 200 meters perhaps. (A major highway passes about a mile from it) and that is placed on an enormous ground screen observed during installation. The point is that it is not movable. It is steered, according to Raytheon, by phasing to the various individual LPA elements in the array. No moving parts, that is.

> Have you attempted (out of interest) to measure interference in front the the LPA's say at a mile or two?

No I havent, but it could be done from that highway mentioned above, which passes right in front of the array a mile or so in front of it. It is interesting to note, that the newspaper indicated that this OHR system was to be used to observe the caribean area. A line drawn on a map from the array to the Caribean passes directly over the FCC Primary Monitoring Station (KI) eight miles away. The monitoring station recently closed, but for unrelated reasons.
If you are interested, I do have a recording of one of the transmissions from the OHR which I made while Raytheon was testing it. Should anyone be interested it could be made into a wave file and uploaded. I posted inquiries to the Brit's radio newsgroup wondering if their signals could be heard on the ham bands there but no one commented.
In conclusion, it should be noted that aside from likely computer

controllers of this system, it is not particularly new technology. OHR
has been around awhile. Considering that, it could be that someone at
Raytheon, the designer and builder, might be willing to contribute
something of interest to this group.

> Do the LPAs point away from built up areas?

        Aside from the highway traffic passing by in front of the array there
are no built up areas in the immediate vicinity. It is wide open Texas
ranching country. I would say that all of the signal passes over the
King Ranch pastures.
>
> So how do the military do it? Are they really using kilowatts of power? How short are their bursts?

        I doubt if those questions would be answered, but considerable power is
available with the typical Harris transmitters. Yes kilowatts.
>
> It would be very interesting to get a wide band receiver (4-30MHz) with
> a fast envelope detector, and very fast peak hold and bring it close to
> the SS instalation in Texas. Surely this would make the TX power
> detectable, allowing comparisons with other TX sources, and giving an
> idea of how the signal is spread.
>
> I expect the military have spent millions on reducing interference (Low
> probability of detection) and they use extreemly wide bandwidths. A
> radar may well have quite different characteristics compared to a
> communications system.
>
> I expect most of their techniques are classified, but there must be
> something we can learn from them. From my tests a simple hopping 2ms
> burst is unacceptable. How do they do it??
>
> My tests suggest that a DSSS system with 10 microwatts power in 2.4kHz
> (100mW in 24MHz) would be detectable at 3 miles. A 10mW carrier 2.4kHz
> SNR is about 0dB (still audible above the noise) at 3 miles using a 70cm
> long unmatched whip antenna - shown to be at least 30dB less sensitive
> than a reasonable amateur station in the presence of ambient noise.
>
> Cheers,
>
> Rob, G0VTQ

41. [HFSIG:1429] Re: SS on HF bands

From: Robert.Glassey@nmp.nokia.com
Date: Mon, 5 Aug 96 16:01:39 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022c0ae2@MHS>
Rendered body type: text/plain

> I guess it boils down to what you think ham radio is all
> about. Reliable communications? That description hardly fits direct
> long-haul HF. Easily accessible to those without large lots, liberal
> housing covenants and money for big antennas and power amplifiers? I
> think not. The popularity of VHF/FM and repeater networks shows there
> are a lot of hams who are interested primarily in reliable local
> communications with small stations, and who aren't purists when it
> comes to using relays.

Very true, but then their operation does not lessen the operation or
experimentation of others. Please remember that experimentation for the
purpose of self training is not just for the expert, but also for the
beginner who wants to build a home brew QRP rig, or get a simple RTTY
hack going, or play around with antennas and station configurations to
achieve the best possible propagation under difficult conditions. The
beginner is not up to the level of high efficency systems, he is more
concerned with the effects of SWR on his transmission line, and why his
low vertical performs better than his low dipole, and how much better a
quad might be and how they work. Trying to work 100 countries is a great
incentive to try and understand these things. This is valid
experimentation, and contributes to the understanding of radio
techniques amounst people who are not 'rocket scientists'. By all means
the experts should challange the new frontiers, but not by treading on
the other folks who have less ambitious aims.

Some have critisized traditional narrow band operaters, claiming that
they consider the whole band to be theirs, but really they only
occasionally claim a few kHz, without upsetting anyone else. They see a
few SS zealoites as attempting to hijack the *entire* band as their own,
causing interference and lessening the enjoyment of everyone else.

> where we trail the commercial and military world. While I don't really
> have anything against DXing and contesting, I really don't see how
> they help justify ham radio's existence to the outside world; they're
> really little more than technological stamp-collecting.

see above.

> And whatever risks there are will be worth the benefits. After all,

Who's cost, who's benifit. Not everyone thinks SS is the 'one true
mode', or needs its efficency. Others have other interests.

By all means experiment but not to the detriment of others, its not your
band either. As they say, 'Fools rush in...' and all that. We can start
on a much smaller scale. This is not an ideal world, only the real one.

My weekend experiments have convinced me that I would not like an SS
operator hopping over the entire 20m band, running more than 100
microwatts within 5 miles of my QTH.

Rob.

42. [HFSIG:1436] Re: HF SS Demonstration

From: Robert.Glassey@nmp.nokia.com
Date: Mon, 5 Aug 96 16:46:43 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022c4002@MHS>
Rendered body type: text/plain

> Odd. I find stations tuning up--even weak heterodynes--a lot more
> annoying than SSB splatter or the occasional solar noise burst. The
> San Diego county noise ordinance lowers the one-hour average SPL noise
> limit by 5 dB if the noise contains a steady, audible tone "such as a
> whine, screech, or hum."

Yeah, I know what you mean. I tried a 2ms burst, 10ms burst and a steady
carrier. The stready carrier was easilly the most audible, and the 10ms
burst was worse than the 2ms burst. Compared to the 2ms burst the noise
burst (30ms) was more annoying.

Cheers,

Rob

43. Re: Amateur culture

From: sivula@NCSNRS01ES.ntc.nokia.com (Sivula Timo (NTC/Espoo))
Date: Mon, 05 Aug 1996 13:56:51 +0300
To: hfsig@tapr.org (hfsig)
Message-ID: <1996Aug05.120248.1604.258127@ntcit-mmta6.ntc.nokia.com>
Rendered body type: text/plain

Phil writes:

>Indeed, ham radio doesn't exist in a vacuum. It is not just a hobby.
>It is given spectrum that could otherwise be sold for big bucks
>because it supposedly gives something back to society as a whole. Now
>that emergency communications and international good will have largely
>been taken over by cellular telephones and the Internet, respectively,
>what's left?

Well, what about its educational influence? I do not know whether this
applies
internationally, please advice me, but in Finland the amateur hobby has
very
often lead starting technical studies. In this way I think that society has
got something
back from the giving away of spectrum. I do not think it is a coinsidence
that many
of our colleagues are hams. The hobby is of course just one of the reasons.
Anyway
there is a correlation between spectrum allocated for amateurs and the
amount of
radio engineers, I think.

An other area where society has gained from the hobby is in defence. In
small countries
that cannot afford an expensive professional army, self educated troops are
very valuable. Very often Radio Amateurs only need little extra training

for
becoming good military radio operators and radio officers.

>Technical experimentation and self-training! How much of that
>really goes on in ham radio these days? Not much

I would like to disagree on this point, as well. There is a lot of self
training and experimenting
going on, on the individual level. A normal HF SSB QSO from Finland to the
USA requires a
lot of know how. Everything is of course well documented and existing old
information,
but for the unexperienced individual newcomer HAM this qso is extremenly
self educating
and it is experimental for him/her, as a person. For me, the assembling of
my 7 MHz QRP
XTAL CW rig, a few years ago, was self educating and experimental. As is
now the testing
and writing of code for the DSP4

IMHO we should not restrict the evaluation of the amateur radio hobby to the
absolute edge
of development only. Other areas can be valuable for the society, as well.

ObHFsig: Is there anybody in the US or outside Europe that would be
interested in testing
SP9VRC´s modem software in a QSO to Finland? The software run on a motorola
EVM or
an Alef Null DSP4. I would be interested in testing the slow BPSK modem and
the faster DQPSK
modems in a DX QSO. Required HW is a HF capable amateur station and an EVM
or DSP4
and a PC.

Has anybody any experience of writing a computer control code for the Yaesu
FT 757 GX?
My code for the DSP4 most probably would gain from hearing some experience
from others...

Timo, OH6KK & OH2LVZ

Timo.Sivula@ntc.nokia.com

44. SS on HF

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Mon, 5 Aug 1996 11:59:53 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <199608051859.AA259141593@polecat.sr.hp.com>
Rendered body type: text/plain

Phil Karn <karn@unix.ka9q.ampr.org> wrote:

>I agree, and actually that's why I would support dividing up the band
>into "DX" and "regional" subbands to mitigate the near-far problem.
>Each subband would still allow wideband modulation within it, but only
>for the right purpose (i.e., DX or local). I think this would go a
>long way towards solving the problem, don't you?

No. There is no way to separate local from DX stations on HF. Local
stations in the "local" band will be drowned out by other "local"
stations thousands of miles away. And DX stations in the "DX" band
will be munged by other DX stations (some close by and some far away).

>... If everyone's power is matched at the receiver, you can tolerate as
>many as 31.6 other users before their aggregate interference brings
>the Eb/N0 down to 6 dB. ...

To assume that everyone's power is the same is a reasonable first
approximation for a VHF cellular system (give or take 10 dB or so).
But on the HF bands, there are huge differences between signal levels.
It would be rare to be working a station that is within 10 dB or so of
the strongest signal on the band.

>>Spread spectrum works fine for something like a cellular telephone system
>>where the locations and power levels of all stations are well-controlled.
>>But on the amateur bands, stations are randomly distributed.
>
>Most cellular users are randomly distributed too. But they do generally
>talk only to the nearest base station.

You obviously know much more about cellular communications than I do,
but as I understand it:

Since uplink and downlink frequencies are separated by 45 MHz, the mobile
stations will not interfere with each other. Since the mobile stations'
power is controlled by the base stations, mobile stations don't interfere
with base stations. And since mobile stations always talk to the closest
base station, base stations don't interfere with mobile stations.

In all cases, interfering signals are never very much stronger than the
desired signals. A cellular system is a carefully-designed system with
tight controls on location and power levels of transmitters. It also
depends on repeatable propagation characteristics. None of those
conditions exist on the HF amateur bands.

>This is true as far as it goes, but 100 dB adjacent channel rejection
>seems a little generous. 60 dB is a more reasonable estimate, I think.
>Whatever.

Good HF communications receivers these days can have 130 dB or more of
blocking dynamic range for interfering signals 20 kHz away. "Adjacent
channel rejection" (3 kHz away?) would be less than that, but 100 dB
seems a pretty conservative typical number. A receiver with only
60 dB of dynamic range would be horrible.

>>Cellular-style communications doesn't work on DX bands, because
>>nearby stations are typically very much weaker than distant stations.
>>(Because ground wave propagation is much lossier than sky wave.)
>
>Ah, but wouldn't this tend to mitigate the near-far problem?

It's not a "near-far" problem so much as a "strong-weak" problem.
On HF, the strongest stations are rarely the closest ones. At a given
time, a station in Tokyo can easily be 60 dB louder than a similar
station in Denver.

>... That means keeping up with the state of the art in the
>non-amateur world, and spread spectrum is certainly one technology
>where we trail the commercial and military world.

I agree with that. But we need to be sure we don't apply commercial
technology inappropriately. To some extent, that's what happened with
AX.25 packet radio. We took a system that was designed for wire LANs
and adopted it with few changes for radio. It works, but not as well
as it should.

AL N1AL

45. Re: [HFSIG:1438] Re: SS on HF bands

From: Greg Jones <wd5ivd@tapr.org>
Date: Mon, 5 Aug 1996 15:33:50 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608052033.PAA01229@tapr.org>
Rendered body type: text/plain

>
> My weekend experiments have convinced me that I would not like an SS
> operator hopping over the entire 20m band, running more than 100
> microwatts within 5 miles of my QTH.
>

Hello.

I don't think you should make a choice between 'good' or 'bad' based on your
overly simplistic and not very scientific experiment.

Doing these types of analogy testing is good and can help people understand
what is at work here -- but there are certain standards required when doing
these types of testing to give them high trustworthiness and creditability.

Cheers - Greg, WD5IVD

46. Re: [HFSIG:1440] Re: Amateur culture

From: Mike Curtis <wd6ehr@kaiwan009.kaiwan.com>
Date: Mon, 5 Aug 1996 14:15:10 -0700 (PDT)
To: hfsig@tapr.org

Message-ID: <199608052115.OAA24677@kaiwan009.kaiwan.com>
Rendered body type: text/plain

> Well, what about <amateur radios> educational influence?

I can't speak about other countries, but in the USA, you only have to memorize test answers to get a license. Unfortunately, that's precisely what MOST new (and upgrading) amateurs do. Amateur radio here is generally not educational - it has been reduced to nothing more than a test of short term memory.

But in any case, education is not part of our rules here, FCC rules part 97. Advancing the state of the art and other technical aspects are the main reasons given for our existence. "International goodwill" and "Trained radio operators" are mentioned once each, out of the 5 reasons given.

-- mike

47. Re: [HFSIG:1442] Re: SS on HF bands

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 5 Aug 1996 15:39:46 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608052239.PAA29230@servo.qualcomm.com>
Rendered body type: text/plain

Greg,

Actually, I think Robert's experiments are a good start. You may quibble with the details of the experiments, but it's hard to argue that at least trying some actual tests can't add some light to the heat of the discussion. They should be encouraged. And I thought simulating just a narrowband "piece" of the spread spectrum signal was a clever idea.

The one comment I have to Robert is this: yes, you may have been able to detect your signals given that you knew they were there and you were specifically looking for them. But being able to simply detect the simulated SS signal (especially on a quiet band) is not the same as saying they'd interfere with your ability to communicate under typical conditions. Would you have been able to detect them had there been a typical SSB signal on the channel in question, and if you had would it have caused any appreciable interference? That's the real issue here.

Phil

48. Re: [HFSIG:1444] Re: SS on HF bands

From: Greg Jones <wd5ivd@tapr.org>
Date: Mon, 5 Aug 1996 18:18:44 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608052318.SAA08883@tapr.org>
Rendered body type: text/plain

I wasn't trying discourage such work -- that just making a judgement to the mode as being good or bad based off of one small test is not correct.

There is more involved, as you point out below, to the entire process.

I hope we can continue to foster such work and development, but not make judgements based on first impressions.

Cheers - Greg

>
> Greg,
>
> Actually, I think Robert's experiments are a good start. You may
> quibble with the details of the experiments, but it's hard to argue
> that at least trying some actual tests can't add some light to the
> heat of the discussion. They should be encouraged. And I thought
> simulating just a narrowband "piece" of the spread spectrum signal was
> a clever idea.
>
> The one comment I have to Robert is this: yes, you may have been able
> to detect your signals given that you knew they were there and you
> were specifically looking for them. But being able to simply detect

> the simulated SS signal (especially on a quiet band) is not the same
> as saying they'd interfere with your ability to communicate under
> typical conditions. Would you have been able to detect them had there
> been a typical SSB signal on the channel in question, and if you had
> would it have caused any appreciable interference? That's the real
> issue here.
>
> Phil
>
>

49. RE: [HFSIG:1443] Re: Amateur culture

From: Jon Bloom <jbloom@connix.com>
Date: Mon, 5 Aug 1996 19:28:37 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB8304.4FA5E2C0@jbloom.connix.com>
Rendered body type: text/plain

Mike Curtis[SMTP:wd6ehr@kaiwan009.kaiwan.com] wrote:
>
>> Well, what about <amateur radios> educational influence?
>
>
>I can't speak about other countries, but in the USA, you only have to
>memorize test answers to get a license. Unfortunately, that's precisely
>what MOST new (and upgrading) amateurs do. Amateur radio here is generally
>not educational - it has been reduced to nothing more than a test of short
>term memory.

The educational aspects of Amateur Radio need not, and do not, end with the taking of the test. The license requirements exist to set minimum standards that ensure (in theory) that the licensee can safely and properly operate a station. Period. (Whether the current exams meet that goal is a subject best left to a forum other than HFSIG.) I suspect that most amateurs learn far more after getting licensed than they did to *get* licensed. I know I did.

I should also add that Amateur Radio is used as an adjunct to formal educational curricula in many schools. Its role in education is not limited to self-training.

>But in any case, education is not part of our rules here, FCC rules part 97.
>Advancing the state of the art and other technical aspects are the main
>reasons given for our existence. "International goodwill" and "Trained
>radio operators" are mentioned once each, out of the 5 reasons given.

As is contributing to the advancement of the state of the art. But that's neither here nor there. Nowhere in Part 97.1 is priority given to any of the reasons cited. Amateur Radio's role as a voluntary noncommercial communication service is no lesser a part of 97.1 than its role in contributing to the state of the art. And if the 97.1 paragraphs that refer to "advancing skills in both the communications and technical phases of the art" and "expansion of... trained operators, technicians, and electronics experts" don't make education part of our rules, what would?

In addition to which, we would be foolish indeed to put all of our eggs in one basket, so to speak. Fulfilling a number of roles gives a much stronger case than relying on one--or one "main"--role as a rationale for our existence.

As it applies to the current HFSIG discussions, I think that part of our educational process should include investigating new techniques and technologies. On that basis, I'd love to see some exploration of SS techniques on HF. But since nowhere in Part 97 is the rationale for Amateur Radio stated as the development of a maximally efficient communication system, it's not a given--at least to me--that we should abandon all narrowband activity in favor of an SS system even if such a system can be shown to be more spectrum-efficient in theory and practice. Which means that any SS system must coexist with narrowband uses of the spectrum by whatever means make that possible.

-- Jon

50. Probability question

From: Richard Lay <lay@cod.nosc.mil>
Date: Mon, 5 Aug 96 16:30:03 PDT
To: hfsig@tapr.org

Message-ID: <9608052330.AA27592@marlin.nosc.mil>
Rendered body type: text/plain

This question has to do with HF ALE radios and their link establishment protocol. One radio sends out a sequence of ALE words. If the receiver finds these words 'appropriate' or 'correct' from its point of view, it will reply, etc. and the link is established. If the receiver detects so many 'inappropriate' or 'incorrect' or 'bad' ALE words in a row, it will disestablish the link and continue scanning. Now, if the probability that one received ALE word is incorrect is = P, and the sequence of ALE words is S long, then what is the probability that _at least_ Y words _in a row_ are incorrect? It's the "at least" and "in a row" that are giving me trouble!

Rich

51. How much QRM is too much?

From: k6sti@n2.net (Brian Beezley)
Date: Mon, 5 Aug 1996 17:35:57 -0700
To: hfsig@tapr.org
Message-ID: <199608060035.RAA13794@ravel.n2.net>
Rendered body type: text/plain

>The one comment I have to Robert is this: yes, you may have been able
>to detect your signals given that you knew they were there and you
>were specifically looking for them. But being able to simply detect
>the simulated SS signal (especially on a quiet band) is not the same
>as saying they'd interfere with your ability to communicate under
>typical conditions. Would you have been able to detect them had there
>been a typical SSB signal on the channel in question, and if you had
>would it have caused any appreciable interference? That's the real
>issue here.

For someone trying to copy a signal right at the thermal, cosmic, or atmospheric noise level, any interference is appreciable. For some hams (particularly 160m, 6m, and VHF/UHF weak-signal types but also for some HF DXers and QRPers), trying to make two-way contact with extremely weak signals constitutes a large part of their amateur activities. These people will never welcome power that could have been put elsewhere, even if it raises their noise floor only by a dB or two. An EMEer can turn blue at the thought of losing of a few tenths of a dB!

Brian Beezley, K6STI
k6sti@n2.net

52. Re: [HFSIG:1447] Probability question

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Mon, 5 Aug 1996 21:21:39 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608060221.VAA11427@www.kelly-afb.org>
Rendered body type: text/plain

Rich,

If you really want the Mil-Std protocol I can get it but I think you are basically correct. There are a couple things that makes ALE not really desirable for hambands...one is that the query transmitter transmitts a _long_ (very long) series of "words" at a very low baud rate...its called "order wire". The length is long enough for the desitnation/desired receiver to scan _all_ the channels in the ALE program. If the "network" has 8-12 channles the scan time for the Mil-Std ALC could be over a minute...this means that the quesy transmitter would transmit for over a minute. Talk about QRM...and to make matters worse, the Mil-Std ALE protocol doesn't listen on a channel before it transmitts...if you have an established communication (QSO) going on, it will walk right on top of you. We disabled it in Saudi during the Gulf War ('90-'91) in favor of doing a daily propagation study using Minimuf...this worked better.

The Mil-Std and Fed-Std have been pushed by several hams thru the ARRL and there was even an big article in QST a couple of years ago.

73, Walt

In your message you write:
>
> This question has to do with HF ALE radios and their link establishment

```
> protocol.  One radio sends out a sequence of ALE words.  If the receiver
> finds these words 'appropriate' or 'correct' from its point of view, it will
> reply, etc. and the link is established.  If the receiver detects so many
> 'inappropriate' or 'incorrect' or 'bad' ALE words in a row, it will
> disestablish the link and continue scanning.  Now, if the probability that
> one received ALE word is incorrect is = P, and the sequence of ALE words is
> S long, then what is the probability that _at least_ Y words _in a row_ are
> incorrect?  It's the "at least" and "in a row" that are giving me trouble!
>
> Rich
>
>


--
=====================================================================
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development.      |
=====================================================================
|                                  |                                |
|                                  | The greatest dangers to liberty |
| Walt DuBose - K5YFW              | lurk in insidious encroachment  |
| E-Mail k5yfw@www.kelly-afb.org   | by men of zeal, well-meaning    |
| Business Telephone: (210)925-6081 | but without understanding.      |
|     Home Telephone: (210)696-3196 |                                |
|                                  |   - Justice Louis D. Brandeis   |
|                                  |                                |
=====================================================================
```

53. Re: [HFSIG:1446] Re: Amateur culture

```
From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Mon, 5 Aug 1996 21:42:23 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608060242.VAA11463@www.kelly-afb.org>
Rendered body type: text/plain
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Jon, et al,

I've remained silent too long on this subject.

I have to agree that most hams do most certainly learn more after they
receive their "first" license.  Few folks in the U.S. go directly from
being un-licensed to Extra Class.  When a ham stops learning, they should
turn in their license.

There is a ham, and perhaps on this SIG, but known to the ARRL Educational
folks (Jon ask Rosalie) who got a big bucks job a with a cellphone company
because of his hands-on "radio" experience through hamradio...even tho
his had not finished his degree...others applying for the job had
degrees, post-grad. degrees, but no hands-on experience.  Our hambands
are becomming the only available frequencies that educational
institutions can readly use for experimentation...so all you teachers,
Middle school thru post grad. get a ham ticket.

I am on the Board of Directors of the Young Astronaut Technology Program
in the Northeast School district here in San Antonio and we use ham radio
as a teaching tool.

One other point.  Many of us here in San Antonio have had the oppertunity
to "take back" event communications from trunking radios and/or cellphones.
The individual with the radio 1) may not be trained and does a bad job of
communiting 2) doesn't listen to the radio 3) Is too busy taking care of
event business to mess with the radio.  Our radio operators are the VIPs
ears and voice.  Ah ha, now I remember why in WWII and the Korean
Conflict we had radio operators...and radio operators on ships and
aeroplanes...the NCOIC Officer, Captain, pilot is too busy making
decisions to listen to the radio.  Also, cellphones don't talk to a whole
bunch of persons all at once.

73,  Walt

 In your message you write:
>
> Mike Curtis[SMTP:wd6ehr@kaiwan009.kaiwan.com] wrote:
> >
> >> Well, what about <amateur radios> educational influence?
> >
> >
> >I can't speak about other countries, but in the USA, you only have to
```

> >memorize test answers to get a license. Unfortunately, that's precisely
> >what MOST new (and upgrading) amateurs do. Amateur radio here is generally
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> The educational aspects of Amateur Radio need not, and do not, end with
> the taking of the test. The license requirements exist to set minimum
> standards that ensure (in theory) that the licensee can safely and
> properly operate a station. Period. (Whether the current exams meet that
> goal is a subject best left to a forum other than HFSIG.) I suspect
> that most amateurs learn far more after getting licensed than they
> did to *get* licensed. I know I did.
>
> I should also add that Amateur Radio is used as an adjunct to formal
> educational curricula in many schools. Its role in education is not
> limited to self-training.
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> >But in any case, education is not part of our rules here, FCC rules part 97.
> >Advancing the state of the art and other technical aspects are the main
> >reasons given for our existence. "International goodwill" and "Trained
> >radio operators" are mentioned once each, out of the 5 reasons given.
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> As is contributing to the advancement of the state of the art. But that's
> neither here nor there. Nowhere in Part 97.1 is priority given to any of
> the reasons cited. Amateur Radio's role as a voluntary noncommercial
> communication service is no lesser a part of 97.1 than its role in
> contributing to the state of the art. And if the 97.1 paragraphs that
> refer to "advancing skills in both the communications and technical
> phases of the art" and "expansion of... trained operators, technicians,
> and electronics experts" don't make education part of our rules, what
> would?
>
> In addition to which, we would be foolish indeed to put all of our eggs
> in one basket, so to speak. Fulfilling a number of roles gives a much
> stronger case than relying on one--or one "main"--role as a rationale
> for our existence.
>
> As it applies to the current HFSIG discussions, I think that part of
> our educational process should include investigating new techniques and
> technologies. On that basis, I'd love to see some exploration of SS
> techniques on HF. But since nowhere in Part 97 is the rationale for
> Amateur Radio stated as the development of a maximally efficient
> communication system, it's not a given--at least to me--that we should
> abandon all narrowband activity in favor of an SS system even if such
> a system can be shown to be more spectrum-efficient in theory and
> practice. Which means that any SS system must coexist with narrowband
> uses of the spectrum by whatever means make that possible.
>
> -- Jon
>
>

--
==
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development. |
==
	The greatest dangers to liberty
Walt DuBose - K5YFW	lurk in insidious encroachment
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Business Telephone: (210)925-6081	but without understanding.
Home Telephone: (210)696-3196	
	- Justice Louis D. Brandeis
==

54. Re: [HFSIG:1442] Re: SS on HF bands

From: Charles Hoffman <choffman@pelican.davlin.net>

Date: Mon, 05 Aug 1996 21:57:22 -0500
To: hfsig@tapr.org
Message-ID: <3206B492.2E70@pelican.davlin.net>
Rendered body type: text/plain

Gentlemen, please! May I make an humble and polite suggestion? After
some 37 years of observations of amateur radio, military and commercial
technological pioneering, and as a former university professor
responsible for raising the awareness of my students-in-ernest, like

yourselves, please permit me to suggest that it is always important to
do your research. See what has already been done. Experiments with
interference is history, check your technical library, do your homework
so to speak and beyond all, do not invent with a thesauraus in your left
hand. All good ideas, but everything I have read here for some time has
already been done. Ask the questions and get the answers, then go from
there. SS is over twenty years here, eh? Perhaps you might check with
W3XO and some of the much earlier pioneers in SS to see what they have
become bored with. HO HO!

Everyones efforts are recognized and appreciated greatly, believe me,
and I am speaking for myself, but I am reading great ideas (seriously)
but which have already been invented. Read your resources and save your
greatest creativity for that which has not yet been tried, concluded;
focus, based upon not what has already been done, but amplify from there
with your refreshing creations of cogitation.

This is meant to be complimentary, but I am becoming somewhat frustrated
with the thread (perhaps, "been there done that", will mean something in
the popular venacular--no offense intended 8-o). Please carry on
gentlemen, you have my undivided attention. We are all one anothers
teachers; I do hope my decades of experience is somehow beneficial as we
are indeed all on the same tangent here. WEll done to all.

73 Rik de K5SBU

Greg Jones wrote:
>
> >
> > My weekend experiments have convinced me that I would not like an SS
> > operator hopping over the entire 20m band, running more than 100
> > microwatts within 5 miles of my QTH.
> >
>
> Hello.
>
> I don't think you should make a choice between 'good' or 'bad' based on your
> overly simplistic and not very scientific experiment.
>
> Doing these types of analogy testing is good and can help people understand
> what is at work here -- but there are certain standards required when doing
> these types of testing to give them high trustworthiness and creditability.
>
> Cheers - Greg, WD5IVD

55. Re: [HFSIG:1449] Re: Probability question

From: Charles Hoffman <choffman@pelican.davlin.net>
Date: Mon, 05 Aug 1996 22:57:40 -0500
To: hfsig@tapr.org
Message-ID: <3206C2B4.64C@pelican.davlin.net>
Rendered body type: text/plain

100% on this one guys...Walt is on target here; forget about ALE on
amrad. First we are not appropriately authorized. Second, the
propogation avenues are not appropriate for ALE on the linear harmonic
related freqs we have available. This system is designed for total HF
access. Lets refocus here.

Walt DuBose - K5YFW wrote:
>
> Rich,
>
> If you really want the Mil-Std protocol I can get it but I think you are
> basically correct. There are a couple things that makes ALE not really
> desirable for hambands...one is that the query transmitter transmitts a
> _long_ (very long) series of "words" at a very low baud rate...its called
> "order wire". The length is long enough for the desitnation/desired
> receiver to scan _all_ the channels in the ALE program. If the "network"
> has 8-12 channles the scan time for the Mil-Std ALC could be over a
> minute...this means that the quesy transmitter would transmit for over a
> minute. Talk about QRM...and to make matters worse, the Mil-Std ALE
> protocol doesn't listen on a channel before it transmitts...if you have an
> established communication (QSO) going on, it will walk right on top of
> you. We disabled it in Saudi during the Gulf War ('90-'91) in favor of
> doing a daily propagation study using Minimuf...this worked better.
>
> The Mil-Std and Fed-Std have been pushed by several hams thru the ARRL
> and there was even an big article in QST a couple of years ago.
>

> 73, Walt
>
> In your message you write:
> >
> > This question has to do with HF ALE radios and their link establishment
> > protocol. One radio sends out a sequence of ALE words. If the receiver
> > finds these words 'appropriate' or 'correct' from its point of view, it will
> > reply, etc. and the link is established. If the receiver detects so many
> > 'inappropriate' or 'incorrect' or 'bad' ALE words in a row, it will
> > disestablish the link and continue scanning. Now, if the probability that
> > one received ALE word is incorrect is = P, and the sequence of ALE words is
> > S long, then what is the probability that _at least_ Y words _in a row_ are
> > incorrect? It's the "at least" and "in a row" that are giving me trouble!
> >
> > Rich
> >
> >
>
> --
> ==
> | The MicroSoft operating system didn't get as bad as it is overnight,|
> | it has taken over 10 years of careful, calculated development. |
> ==
> | | |
> | | The greatest dangers to liberty |
> | Walt DuBose - K5YFW | lurk in insidious encroachment |
> | E-Mail k5yfw@www.kelly-afb.org | by men of zeal, well-meaning |
> | Business Telephone: (210)925-6081 | but without understanding. |
> | Home Telephone: (210)696-3196 | |
> | | - Justice Louis D. Brandeis |
> | | |
> ==

56. Re: [HFSIG:1446] Re: Amateur culture

From: Mike Curtis <wd6ehr@kaiwan009.kaiwan.com>
Date: Tue, 6 Aug 1996 00:14:34 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608060714.AAA17899@kaiwan009.kaiwan.com>
Rendered body type: text/plain

>
> Mike Curtis[SMTP:wd6ehr@kaiwan009.kaiwan.com] wrote:
> >
> >> Well, what about <amateur radios> educational influence?
> >
> >
> >I can't speak about other countries, but in the USA, you only have to
> >memorize test answers to get a license. Unfortunately, that's precisely
> >what MOST new (and upgrading) amateurs do. Amateur radio here is generally
> >not educational - it has been reduced to nothing more than a test of short
> >term memory.
>
> The educational aspects of Amateur Radio need not, and do not, end with
> the taking of the test. The license requirements exist to set minimum
> standards that ensure (in theory) that the licensee can safely and
> properly operate a station. Period. (Whether the current exams meet that
> goal is a subject best left to a forum other than HFSIG.) I suspect
> that most amateurs learn far more after getting licensed than they
> did to *get* licensed. I know I did.
>
> I should also add that Amateur Radio is used as an adjunct to formal
> educational curricula in many schools. Its role in education is not
> limited to self-training.

I agree that a lot of hams learn more after being licensed than to be
licensed. But my points were that 1. to GET the license requires ONLY rote
memorization, and 2. unfortunately MOST hams never progress much beyond what
is required.

Both of us have taken the lead in education among hams. And there have been
(and will be) a lot of hams who are interested in learning what they really
should know for the license - and beyond. Unfortunately, many of the
ignoramuses who are just interested in appliance operation make things
difficult. There used to be a popular repeater here in LA where technical
talk was "officially" discouraged. I don't know if it's still "no-tech", or
even if it's still on the air, but that's unimportant.

> As it applies to the current HFSIG discussions, I think that part of
> our educational process should include investigating new techniques and

> technologies. On that basis, I'd love to see some exploration of SS
> techniques on HF. But since nowhere in Part 97 is the rationale for
> Amateur Radio stated as the development of a maximally efficient
> communication system, it's not a given--at least to me--that we should
> abandon all narrowband activity in favor of an SS system even if such
> a system can be shown to be more spectrum-efficient in theory and
> practice. Which means that any SS system must coexist with narrowband
> uses of the spectrum by whatever means make that possible.

One of the best things about amateur radio IMHO is its diversity. I think most of us will agree with this wholeheartedly. Nowhere have I seen a proposal to pursue any particular mode to the exclusion of all others. So I don't see abandonment of all narrowband technology as anything to be concerned about.

Over the years, we've had to make adjustments in band usage. When RTTY became more popular, we made room for it. When SSTV came around, we adjusted. Ditto packet, AMTOR, et al. But all the "old" modes are still around (well, except spark, but that's not the fault of anything other than the mode itself :-)

One thing for sure - it's going to be most interesting watching things unfold.

-- mike

57. EVM56002 ?

From: Greg Jones <wd5ivd@tapr.org>
Date: Tue, 6 Aug 1996 04:12:31 -0500 (CDT)
To: hfsig@tapr.org (HF SIG mailing)
Message-ID: <199608060912.EAA05283@tapr.org>
Rendered body type: text/plain

There is an opportunity for TAPR to do a group purchase on the Motorola EVM56002 at a pretty low cost. Between 20 and 200 units would be the goal...same price no matter how many are purchased. Figure about $60 under current commercial prices. No hard prices, becuase more work would have to be done.

Would there be interest in this in the HF SIG ? Espiecally since Johan's code would run on it.

I have spoken with Johan and he thinks it would be great...what are some other reactions.

Cheers - Greg, WD5IVD

58. [HFSIG:1442] Re: SS on HF bands

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 6 Aug 96 10:16:32 +0100
To: hfsig@tapr.org, wd5ivd@tapr.org
Message-ID: <H0000292022d7982@MHS>
Rendered body type: text/plain

> >
> > My weekend experiments have convinced me that I would not like an SS
> > operator hopping over the entire 20m band, running more than 100
> > microwatts within 5 miles of my QTH.
> >
>
> Hello.

> I don't think you should make a choice between 'good' or 'bad' based
> on your overly simplistic and not very scientific experiment.

Err, wait a moment, I'll post my report today hopefully, it certianly isn't the be all to end all of propagation studied, but a considerable amount of effort was put into this, and extra special attention was paid to minimising the level of interference a SS signal could generate, including trying various different pulse shapes and power levels.

> Doing these types of analogy testing is good and can help people
> understand what is at work here -- but there are certain standards
> required when doing these types of testing to give them high
> trustworthiness and creditability.

I hope that my experiments have been done to a standard that will give

the results a respectible degree of trustworthyness and credibility.
Considerable effort was put into calibration and characterisation of
both TX and RX. There is of course a large error margin (either way) on
the results (mostly due to the terain), however this was qualified and
considered in context of the severity of the interference observed and
enviromental factors.

The report will be out soon.

Cheers,

Rob

59. [HFSIG:1444] Re: SS on HF bands

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 6 Aug 96 10:37:50 +0100
To: hfsig@tapr.org, karn@qualcomm.com
Message-ID: <H0000292022d798d@MHS>
Rendered body type: text/plain

> The one comment I have to Robert is this: yes, you may have been able
> to detect your signals given that you knew they were there and you
> were specifically looking for them. But being able to simply detect
> the simulated SS signal (especially on a quiet band) is not the same
> as saying they'd interfere with your ability to communicate under
> typical conditions.

Yes, listening for the signal does make it easier to hear, but I did not
consider 'maybe' interference significant, only when clear tone or noise
bursts could be heard. While you may not notice this when briefly
listening to a signal, any such signals will become much more apparent
when you listen for a longer time, especially if you are listening to a
weak station. Typical conditipons are not relevant, since this is not
the area of concern, rather the concern is over the interference caused
to signals that are already dificult to hear, caused by a hopping
transmitter within a few miles.

> Would you have been able to detect them had there
> been a typical SSB signal on the channel in question, and if you had
> would it have caused any appreciable interference? That's the real
> issue here.

As it happens, 18MHz was quite an active band during my tests, and there
were times when even on my 70cm long whip antenna a strong SSB signal
came through strongly. In my tests I compared a constant carrier level
with amateur and broardcast station levels, and pulsed interference with
a constant carrier. My scanner with the whip antenna was not capable of
receiving weak signals, being approximatly 30dB less sensitive that my
dipole at home, in the presence of external noise. Still, the
interference could be heard although very strong amateur stations could
overpower it.

More details in the report...

Cheers,

Rob

60. Re: [HFSIG:1454] EVM56002 ?

From: Mike Murphree <n4cnw@pig.net>
Date: Tue, 06 Aug 1996 07:47:08 -0400
To: hfsig@tapr.org
Message-ID: <320730BC.1500@pig.net>
Rendered body type: text/plain

Greg Jones wrote:

> There is an opportunity for TAPR to do a group purchase on the Motorola
> EVM56002 at a pretty low cost. Between 20 and 200 units would be the
> goal...same price no matter how many are purchased. Figure about $60 under
> current commercial prices. No hard prices, becuase more work would have to
> be done.
>
> Would there be interest in this in the HF SIG ? Espiecally since Johan's
> code would run on it.

Sounds great considering the price is already only $149.95 at the
Design-Net price page (http://www2.motorola-dsp.com/dsp/home/net/pg.html).

They also offer a student discount of 20% reducing the price to $120 + $3
shipping (http://www2.motorola-dsp.com/dsp/home/edu/discount.html)

Mike N4CNW

61. Amateur culture and not so wide modes

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 6 Aug 96 16:04:00 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022d7998@MHS>
Rendered body type: text/plain

Hi Mike, good to hear another voice.

> I agree that a lot of hams learn more after being licensed than to be
> licensed. But my points were that 1. to GET the license requires ONLY
> rote memorization, and 2. unfortunately MOST hams never progress much
> beyond what is required.

True, but I have no problem with this. Nobodys lost out even if they
never pick up a mic. At least they have explored an interest, even
learnt something about radio, however little. Even appliance operators
can get the itch to try something more, to get a bigger antenna, to try
and work the repeater 50 miles away to talk to their friends over there,
for many that small interest can develope into much more. One of the
great things about amateur radio is that there is always more to try, no
matter how far you get involved there is always more chalanges,
something new and interesting. Amateur radio encourages an understanding
of radio in the wider community, to lift some of the mystery of this
'black art'. It might even help people understand how to tune their
video receoders if their friend or relative is a ham :-) But seriously,
ham radio is for everyone, expert, beginner, emergency operators, and
even the appliance operator.

> One of the best things about amateur radio IMHO is its diversity. I
> think most of us will agree with this wholeheartedly. Nowhere have I
> seen a proposal to pursue any particular mode to the exclusion of all
> others. So I don't see abandonment of all narrowband technology as
> anything to be concerned about.

I hope not, just to reprint the few lines that started this thread off:

On 24th July Phil wrote:

> But it's precisely those 300 baud/1 KHz restrictions that I want to
> eliminate. Eventually I want to spread over the whole band with
> frequency hopping, but to start I'd like to do MFSK over SSB
> bandwidths.

This was said in the context of 'how much bandwidth can we hop over' and
the opinions and attitudes that Costas called 'Narrow Band Thinking' in
his classic paper (which I'm sure Phil has framed on his wall :-).

The idea here was simply to use an SSB channel. The previous discussion
involved whether narrow or wide modes were better.

I personally go along with Phil to an extent. I like the idea of trying
this mode out and developing the technology it uses. But while there may
be gains in spreading over the whole band, those gains are for only a
few, and may well be to the detriment of all other users. I am in favour
of say a 10-25KHz allocation for all wide band channel sharing modes. I
see considereable scope here for experimentation, and to overcome the
problems of HF packet. (CSMA sharing using 300 baud on HF would be
hillarious if people didn't actually try to use it!) It needs to be
replaced, and this may be the answer. We need to get clever on this one,
and not just rely on spreading to overcome interference to the users of
this mode by lumping it on everyone else. I think the answer lies in
coding and inteligent use of redundancy, combined with a small degree of
hopping to use the trasciever filters to counter the few very strong
signals (make them random 'hits'). 25kHz would be heaps, and if you dont
need to worry about sharing with other modes, then you have much more
freedom to try different techniques, without the constraints of
non-interference to narrow band DX stations.

This could be thought of as not so much 'spread spectrum' but more a
wide Code Division Multiple Access (CDMA) mode to replace the current
wide (total spectrum usage) carrier sense (CSMA) HF packet system.
(shades of "it's not a bomb" for those few New Zealand readers :-)

Sorry Phil, I think maybe I've overreacted a bit here, especially since your original comment was more to do with lifting the 1kHz 300 Baud restrictions in the US, and using an SSB channel, which I would support anyhow. Its spreading over the whole band that I and many others oppose. Maybe we've come full circle here, but I think a few good comments have come out.

Cheers,

Rob

62. Re: [HFSIG:1457] Re: EVM56002 ?

From: forrerj@peak.org (Johan Forrer.)
Date: Tue, 06 Aug 1996 08:31:19 -0800
To: hfsig@tapr.org
Message-ID: <199608061542.IAA23152@PEAK.ORG>
Rendered body type: text/plain

Hi All,

Let me just recap what the possibilities are:

- A 60 MHz 56002, concervatively rated. Some have shown to run reliably at 80 MHz. (When executing multiple instructions per clock tick, this effectively translates
  to 60*3=180 MIPS or at a minimum, one instruction per clock tick - 60 MIPS).
  This is a 24-bit machine (120 dB dynamic range); for comparison, its TWICE as fast as the 16-bit DSP chip as used in the new Pactor II modem.

- The 56K instruction set looks a lot like Motorola's other microcontrollers, however, is a dual Harward machine and a true DSP. It's a pleasure to write code
  for it.

- Has an on-board serial peripheral, some spare I/O lines and adequate memory for
  doing innovative programming projects.

- Comes with a very decent, full-screen, multiple windows, debugger that was developed
  by Domain Technologies. The debugger software is stable and has undergone several
  revisions giving you the benefit of user-feedback. The debugger is OnCe-based, i.e.,
  uses special on-chip silicon to do non-intrusive debugging at full DSP speeds.

- A decent assembler with macro facilities. Public domain C compiler, linker, librarian and simulator is available.

- A port of the Alef Null monitor "Leonid" bridges the EVM and the Finnish DSP Card 4.
  This gives you access to some outstanding software efforts.

- A substantial amount of amateur radio-related software including modems for packet,
  WEFAX, satellite work, and of course HF digital. To get on the air, you need the EVM and a PC that talks the KISS protocol. Several options are possible.

I own two EVM's myself that have been adapted for amateur radio use. One is configured as a single port radio modem and the other as a dual-port modem - it has expanded memory to 32K words in each of the P, X, and Y memory spaces. When I find the time, I hope to finish an article on the interface for this little beast. One of my future projects is to port the popular multitasking OS called uC O/S (micro-C O/S) to this expanded EVM. It would greatly simplify the development of multithreaded code, especially the kind of thing that is needed when implementing communications protocols. The EVM has plenty of spare horsepower to handle such a task in addition to real-time modems.

Anyone is interested in learning about DSP and am ready to tinker with it at the hardware level, this is a great opportunity. Be assured that the parts on the EVM alone is worth more than TAPR's asking price.

Forgive me if I sound like a Motorola sales person - I don't work for them, just am excited about this bit of hardware and want to make sure that you don't miss up on a good opportunity.

--Johan, KC7WW

>Greg Jones wrote:
>
>> There is an opportunity for TAPR to do a group purchase on the Motorola
>> EVM56002 at a pretty low cost.   Between 20 and 200 units would be the
>> goal...same price no matter how many are purchased. Figure about $60 under
>> current commercial prices. No hard prices, becuase more work would have to
>> be done.
>>
>> Would there be interest in this in the HF SIG ? Espiecally since Johan's
>> code would run on it.
>
>Sounds great considering the price is already only $149.95 at the
>Design-Net price page (http://www2.motorola-dsp.com/dsp/home/net/pg.html).
>
>They also offer a student discount of 20% reducing the price to $120 + $3
>shipping (http://www2.motorola-dsp.com/dsp/home/edu/discount.html)
>
>Mike N4CNW
>
>

63. Re: [HFSIG:1447] Probability question

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)
Date: Tue, 6 Aug 96 13:03:31 CDT
To: hfsig@tapr.org
Message-ID: <9608061803.AA01687@eagle.aud.alcatel.com>
Rendered body type: text/plain

> This question has to do with HF ALE radios and their link establishment
> protocol. One radio sends out a sequence of ALE words. If the receiver
> finds these words 'appropriate' or 'correct' from its point of view, it will
> reply, etc. and the link is established. If the receiver detects so many
> 'inappropriate' or 'incorrect' or 'bad' ALE words in a row, it will
> disestablish the link and continue scanning. Now, if the probability that
> one received ALE word is incorrect is = P, and the sequence of ALE words is
> S long, then what is the probability that _at least_ Y words _in a row_ are
> incorrect? It's the "at least" and "in a row" that are giving me trouble!
>
> Rich
>
>

Rich: I did this rather quickly, but here's a guess:

the probability of Y& words in a row are incorrect should be:

$$(1 - P^Y)^{(S+1-Y)} \qquad (1)$$

the probability of Y or more words in a row are incorrect should be:

$$\sum_{k=0}^{(S-Y)} (1-P^{[Y+k]})^{(S+1-Y+k)} \qquad (2)$$

Again, I have not spent the time to verify this. If P is
small, then (1) may be sufficiently accurate.

- Tom

Tom McDermott
mcdermot@aud.alcatel.com

64. Pulses carrier propagation tests

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 6 Aug 96 19:26:18 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022ed98b@MHS>

Short Distance Ground Wave Propagation of Low Power Pulsed HF Transmissions

Introduction

A series of tests were conducted to determine the propagation characteristics of low power pulsed transmissions on 18 MHz over a distance of 1 to 5 miles. The aim of this investigation was to determine the level of interference a frequency hopping spread spectrum transmitter would cause to nearby stations.

The Transmitter:

A Yaesu FT890AT transceiver was used, with the RF power control set to minimum. This gave an output power of about 3 Watts in CW mode, and the same in LSB mode when the audio drive was sufficient to cause the ALC to operate.
Power was further controlled by adjusting the audio input level in LSB mode. In this way output power was varied between 1 Watt and 10 mW pep. LSB was used since carrier suppression was best in this mode. The carrier was undetectable at 1/2 a mile using the receiver below.

The Built in ATU was used to tune the TX antenna. The TX antenna was an 80m dipole, 4m above the ground. Other amateur station may well use better antennas for transmission.

The audio source was an 8 bit PC sound card.

The Receiver:

Yupiteru MVT7100 handheld scanner, with a 1m whip antenna, or a simple dipole and balun resonating around 18MHz.

Using the unmatched 1m whip antenna received signal levels were 10dB below the dipole with a 20dB attenuator, in the same position. When the dipole was used the MVT7100's built-in 20dB attenuator had to be used due to the poor intermod performance of the receiver. This the whip was 30dB less sensitive than the dipole.

To account for the external noise floor the SNR of weak signals using the MVT7100 + dipole + 20dB attenuator was compared with the FT890 + same dipole. The SNR from the MVT7100 was about 10dB worse. i.e. the 20dB attenuator results in a 10dB drop in SNR since the noise floor from the dipole is about 10dB above the MVT7100 noise floor without the attenuator. Thus the noise floor drops only 10dB while the signal drops 20dB. The SNR of a typical DX setup with directional antennas (lower noise) high off the ground (stronger signal) is probably about 10 dB or better than the low RX dipole I used. Thus the MVT710 + low dipole + 20dB attenuator is about 10+10 = 20 dB worst than a typical DX setup.

The whip antenna was found to be 10dB worst than the dipole + 20dB attenuator. Since SNR with the attenuator and dipole was not limited by the external noise floor, the whip antenna has 30dB lower SNR than a typical DX setup.

The Terrain:

The surrounding area is hilly, part pine forest, and part a lightly wooded common with some rough open areas. Total wooded area is around 20 square miles. The hills are formed by a the erosion of a plateau a few remaining parts of the flat plateau at the hill tops. The plateau is about 120-130m ASL. The site is 30 miles inland. The TX site is on the edge of the wooded land in a built up area, 121m ASL.

The Tests:

CW tests were done with 3 Watts, on 18.090 MHz. Only he whip was used to determine the extent and characteristics of propagation in this area. The signal was compared with the strength of amateur stations and broadcast stations.

Pseudo Frequency Hopping Spread Spectrum tests were done by transmitting a series of pulsed a carriers at random frequencies and times to simulate a Frequency Hopping Spread Spectrum signal as it would appear in a 2.4 kHz bandwidth. The pulses were generated using shaped audio tone bursts to modulate the transmitter in LSB mode. The LSB suppressed carrier frequency was 18.150 MHz.

A series of different PFHSS tests were done trying different PEP power levels and pulse shapes. A constant carrier was also transmitter for reference at each power level.

Four pulse shapes were used, these are described below.

The timing of the pulses followed a Poisson distribution. The mean pulse rate varied depending upon the test.

Power levels were: 1 Watt, 100mW, 10mW, and a CW ID for each test at 3 Watts.

Pulse 1: Square 2ms pulsed tones, random phase and audio frequency 300-2700Hz. Mean pulse rate was 3.4 pulses per second. (equivilant to frequency hopping spread spectrum as appears in a 2.4kHz bandwidth when spread over 350kHz with a 2ms dwell time.)

Pulse 2: Raised cosine pulsed tone, 4ms total, -6db pulse duration 2ms. Random phase and frequency with the same pulse positioning as 1.

Pulse 3: 10ms pulsed tones, raised cosine edge shaping, 2ms each side, 10ms total pulse length. FHSS equivilant hopping rate would be 1.4 seconds per pulse. The pulse rate used was twice this, so enough pulses could be heard to determine the signal level. Audio frequency and phase were random.

Pulse 4: 30ms modulated pulsed tones. No shaping, but random BPSK modulation at 2000 bps. Equivilant FHSS pulse rate would be 4.4 seconds per pulse. The rate used was 1.1 seconds per pulse. Centre frequency and phase were random.

Each test lasted 20 seconds. The computer was set up to sequence through the four pulse tests plus a CW test, at each power level. Each test and power level had a unique morse ID.

The Results:

CW measurements.

The 3 watt CW signal was easily heard in all locations with the MVT7100 and whip antenna beyond 5 miles. At 5 miles the signal was just audible with the whip antenna and the 20 dB attenuator on.

Typical signal levels at 1 to 1.5 miles on in good DX locations (on hill tops at the same level as the TX) were 10-30dB above the strongest amateur stations on 18MHz and 14MHz (using the attenuator and calibrated S meter), and 10-30dB below the strongest broadcast stations (BBC, Deutchewelle, VOA, VORussia, Radio Canada) The signal strength was effected more by terrain than distance. The strongest broadcast stations were typically 30-40dB above the strongest amateur stations. Tests were done between 8pm and 10pm local daylight saving time (GMT+1) on Friday 2/8/96.

Pulsed tests.

SOUND: Pulses 1,2,3 all had discernible tones. Pulse 1 was least discernible, pulse 2 was slightly more pronounced, pulse 3 had a clearly discernible tone. The modulated tone, pulse 4, sounded like a burst of noise, with a discernible beginning, end and pitch. Pulse streams 1 and 2 sounded like many taps dripping water into many buckets, all with different pitches. Pulse stream 3 were clear tone bursts. On air any of these pulses were easily distinguishable from a static click and would be annoying however infrequent.

AUDIBILITY: Pulses 1 and 2 seemed to be about the same loudness in the presence of noise, and were the softest of the four pulses. Pulse 3 was more audible, and pulse 4 was the clearest. In terms of signal strength, pulses 1 and 2
required 20dB more power than a CW signal to be audible. Pulses 3 and 4 required 10dB more power. (these results came from comparing received

signals with different TX powers).

RANGE:

RX measurements were made between 1 and 3 miles from the TX, with both the whip and dipole antenna. Again signal strength depended more upon terrain than distance, although there was a clear difference between 1 and 3 miles. Distances were measure using an Ordinance Survey Map 1:50,000. (standard British maps) These are pretty good maps.

At good receiving sites (on flat ground or hill tops on the same level as the TX) all tests were audible with 1 Watt TX power and the whip antenna at a range of 1-3 miles.

At some sites the 100mW pulse 1 and 2 tests could not be heard with the whip antenna, but tests 3 and 4 could be heard. Using the dipole antenna and the 20dB attenuator often made the pulse 1 and 2 tests audible in these cases.

At 3 miles only the 10mW carrier could be heard, no other 10mW tests could be heard with the whip antenna.

At 1 mile the 100mW pulse 1 and 2 tests could often be heard with the whip antenna. In one case at a distance of 1 mile, the 10mW pulse 1 and 2 tests were clearly audible with the whip antenna.

Since whip antenna was estimated to be more than 30dB less sensitive than a good amateur station when the external noise floor is considered (see above), thus if 1 watt TX power is heard using the whip antenna then 1mW will be heard by a good amateur station. In the case where 10mW pulse 1 and 2 TX power was heard clearly at 1 mile, a good amateur station would hear a 10uW 2ms carrier burst.

Conclusion:

Signal strengths of a 3 Watt CW transmitter at a distance of 1 mile were considerable greater (10-30dB) than any other amateur stations heard on the band.

Shorter pulse lengths are less audible, despite being more frequent.

Attempting to spread a long (30ms) tone to make it less noticeable did not work.

A 1mW 2ms pulse at 18MHz would be clearly audible above the ambient noise floor by a good amateur station up to 3 miles from the transmitter site.

A 10uW 2ms pulse at 18MHz may well be heard by a good amateur station at a distance of 1 mile.

Interference may be more severe if the TX station has a better antenna than a 80m dipole 4m above the ground.

Possible Further Work:

Interference decreased as the pulse length got shorter, and subjectively the sound was less was annoying. If pulse length was made even shorter, say 0.1mS, then the subjective effect of the pulse may be acceptable. Such a short pulse length would spread power wider than a single SSB channel, further reducing the level of the received pulse, and also making it more compatible with common impulse noise blankers. If such a short dwell time were used in a hopping system, spreading over 350kHz, then the average pulse rate in a narrow receiver bandwidth (<3kHz) would be:

$$\text{Rate} = \frac{\text{pulse\_band\_width}}{\text{total\_band}} / \text{dwell\_time}$$

$$= 20 / 350 / 100\text{E-}6$$

$$= 571 \text{ pulses per second on average.}$$

This would probably be unacceptable, since the average energy in the receiver bandwidth would be high and not easily dealt with. This would probably be very similar to a direct sequence system.

If spread over 15MHz, rate = 13 pulses per second on average.

The ideal dwell time (pulse length) would probably be in the range 0.1
to 0.5 ms. Further investigation would be required.

Further reduction in pulse rate could be achieved by trading off more
power for fewer pulses, if the pulses are short enough to sound
sufficiently noise like.

Interference from such a system may be acceptable.

65. Re: [HFSIG:1459] Re: EVM56002 ?

From: zs6awk@global.co.za (Danie Brynard)
Date: Tue, 06 Aug 1996 22:38:46 +0200
To: hfsig@tapr.org
Message-ID: <199608070036.WAA21477@lin01.global.co.za>
Rendered body type: text/plain

I can only agree: the DSP56002EVM is a wonderfull piece of hardware. I have
been in love with mine for 1.5 years :-)

Newcomers please see the software list on the wooster.hut.fi site in Finland
and Johan's compenduim on the TAPR web site.

cu es 73 de Danie zs6awk

>Hi All,
>
>Let me just recap what the possibilities are:
>
>- A 60 MHz 56002, concervatively rated. Some have shown to run reliably at
>80 MHz.   (When executing multiple instructions per clock tick, this
>effectively translates
> to 60*3=180 MIPS or at a minimum, one instruction per clock tick - 60 MIPS).
> This is a 24-bit machine (120 dB dynamic range); for comparison, its
TWICE as
> fast as the 16-bit DSP chip as used in the new Pactor II modem.
>
>- The 56K instruction set looks a lot like Motorola's other microcontrollers,
> however, is a dual Harward machine and a true DSP. It's a pleasure to
>write code
> for it.
>
>- Has an on-board serial peripheral, some spare I/O lines and adequate
>memory for
> doing innovative programming projects.
>
>- Comes with a very decent, full-screen, multiple windows, debugger that was
>developed
> by Domain Technologies. The debugger software is stable and has undergone
>several
> revisions giving you the benefit of user-feedback. The debugger is
>OnCe-based, i.e.,
> uses special on-chip silicon to do non-intrusive debugging at full DSP
speeds.
>
>- A decent assembler with macro facilities. Public domain C compiler, linker,
> librarian and simulator is available.
>
>- A port of the Alef Null monitor "Leonid" bridges the EVM and the Finnish
>DSP Card 4.
> This gives you access to some outstanding software efforts.
>
>- A substantial amount of amateur radio-related software including modems
>for packet,
> WEFAX, satellite work, and of course HF digital. To get on the air, you
>need the   EVM and a PC that talks the KISS protocol. Several options are
>possible.
>
>
>I own two EVM's myself that have been adapted for amateur radio use. One is
>configured as a single port radio modem and the other as a dual-port modem -
>it has expanded memory to 32K words in each of the P, X, and Y memory
>spaces. When I find the time, I hope to finish an article on the interface
>for this little beast. One of my future projects is to port the popular
>multitasking OS called uC O/S (micro-C O/S) to this expanded EVM. It would
>greatly simplify the development of multithreaded code, especially the kind
>of thing that is needed when implementing communications protocols. The EVM

>has plenty of spare horsepower to handle such a task in addition to
>real-time modems.
>
>Anyone is interested in learning about DSP and am ready to tinker with it at
>the hardware level, this is a great opportunity. Be assured that the parts
>on the EVM alone is worth more than TAPR's asking price.
>
>Forgive me if I sound like a Motorola sales person - I don't work for them,
>just am excited about this bit of hardware and want to make sure that you
>don't miss up on a good opportunity.
>
>
>--Johan, KC7WW
>
>
>>Greg Jones wrote:
>>
>>> There is an opportunity for TAPR to do a group purchase on the Motorola
>>> EVM56002 at a pretty low cost.   Between 20 and 200 units would be the
>>> goal...same price no matter how many are purchased. Figure about $60 under
>>> current commercial prices. No hard prices, becuase more work would have to
>>> be done.
>>>
>>> Would there be interest in this in the HF SIG ? Espiecally since Johan's
>>> code would run on it.
>>
>>Sounds great considering the price is already only $149.95 at the
>>Design-Net price page (http://www2.motorola-dsp.com/dsp/home/net/pg.html).
>>
>>They also offer a student discount of 20% reducing the price to $120 + $3
>>shipping (http://www2.motorola-dsp.com/dsp/home/edu/discount.html)
>>
>>Mike N4CNW
>>
>>
>
>

66. Re: [HFSIG:1459] Re: EVM56002 ?

From: forrerj@peak.org (Johan Forrer.)
Date: Tue, 06 Aug 1996 14:58:31 -0800
To: hfsig@tapr.org
Message-ID: <199608062209.PAA00762@PEAK.ORG>
Rendered body type: text/plain

>Hi All,
>
>Let me just recap what the possibilities are:
>
>- A 60 MHz 56002, concervatively rated. Some have shown to run reliably at
>80 MHz.   (When executing multiple instructions per clock tick, this
>effectively translates
> to 60*3=180 MIPS or at a minimum, one instruction per clock tick - 60 MIPS).
> This is a 24-bit machine (120 dB dynamic range); for comparison, its
TWICE as
> fast as the 16-bit DSP chip as used in the new Pactor II modem.

Correction: MIPS factor is actually CLOCK/2, thus the 66 MHz chip
runs at 33 MIPS and the 80 MHz versions at 40 MIPS.

Sorry about that.

--Johan

>
>- The 56K instruction set looks a lot like Motorola's other microcontrollers,
> however, is a dual Harward machine and a true DSP. It's a pleasure to
>write code
> for it.
>
>- Has an on-board serial peripheral, some spare I/O lines and adequate
>memory for
> doing innovative programming projects.
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>- Comes with a very decent, full-screen, multiple windows, debugger that was

>developed
> by Domain Technologies. The debugger software is stable and has undergone
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> revisions giving you the benefit of user-feedback. The debugger is
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>- A decent assembler with macro facilities. Public domain C compiler, linker,
> librarian and simulator is available.
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>DSP Card 4.
> This gives you access to some outstanding software efforts.
>
>- A substantial amount of amateur radio-related software including modems
>for packet,
> WEFAX, satellite work, and of course HF digital. To get on the air, you
>need the   EVM and a PC that talks the KISS protocol. Several options are
>possible.
>
>
>I own two EVM's myself that have been adapted for amateur radio use. One is
>configured as a single port radio modem and the other as a dual-port modem -
>it has expanded memory to 32K words in each of the P, X, and Y memory
>spaces. When I find the time, I hope to finish an article on the interface
>for this little beast. One of my future projects is to port the popular
>multitasking OS called uC O/S (micro-C O/S) to this expanded EVM. It would
>greatly simplify the development of multithreaded code, especially the kind
>of thing that is needed when implementing communications protocols. The EVM
>has plenty of spare horsepower to handle such a task in addition to
>real-time modems.
>
>Anyone is interested in learning about DSP and am ready to tinker with it at
>the hardware level, this is a great opportunity. Be assured that the parts
>on the EVM alone is worth more than TAPR's asking price.
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>Forgive me if I sound like a Motorola sales person - I don't work for them,
>just am excited about this bit of hardware and want to make sure that you
>don't miss up on a good opportunity.
>
>
>--Johan, KC7WW
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>>Greg Jones wrote:
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>>> There is an opportunity for TAPR to do a group purchase on the Motorola
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>>Design-Net price page (http://www2.motorola-dsp.com/dsp/home/net/pg.html).
>>
>>They also offer a student discount of 20% reducing the price to $120 + $3
>>shipping (http://www2.motorola-dsp.com/dsp/home/edu/discount.html)
>>
>>Mike N4CNW
>>
>>
>
>

67. Re: [HFSIG:1440] Re: Amateur culture

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 6 Aug 1996 20:46:11 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608070346.UAA18246@servo.qualcomm.com>
Rendered body type: text/plain

>Well, what about its educational influence? I do not know whether this
>applies internationally, please advice me, but in Finland the amateur
>hobby has very often lead starting technical studies. In this way I
>think that society has

This is exactly my point. In my opinion, the climate toward technical education and experimentation in at least the US amateur world has been rather hostile, as evidenced by the attitudes toward the no-code license and spread spectrum and the tiny fraction of hams doing serious technical experimentation. Not to mention the fact that amateur technology is seriously behind the state of the art. And then you have the armchair lawyers that always seem to appear out of nowhere to claim that anything even slightly new and novel has got to violate some FCC rule somewhere, usually (but not always) the prohibition against codes and ciphers.

Ragchewing, contesting and DXing seem to be 99.9999% of ham radio. While I don't really object to these, it's hard to argue that they carry much weight with the non-amateur public that owns our spectrum.

The points you mention about experimentation, the training of operators and the like are specifically listed in the US amateur rules. Actually, I do agree that ham radio serves a valuable purpose even if the technical experimentation isn't leading-edge; just advancing one's own personal skills are enough. But even with this low standard, only a minority of the amateur population really take advantage of the opportunity. The rest engage in what can best be described as electronic stamp-collecting.

Phil

68. Re: [HFSIG:1460] Re: Probability question

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 6 Aug 1996 21:33:19 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608070433.VAA18353@servo.qualcomm.com>
Rendered body type: text/plain

Tom McD's answer:

>the probability of Y& words in a row are incorrect should be:

>   (1 - P^Y)^(S+1-Y)   (1)
[...]

I get a somewhat different answer. Or at least I followed a different analysis.

The probability that the very first Y symbols in the block of S (with S >= Y) are correct (with individual probability of symbol error P) is

        (1-P)^Y

Now add to this the probability that the first symbol is in error, but the next Y consecutive symbols are correct:

        P * (1-P)^Y

And the probability that the first TWO symbols are in error, but the next Y are correct (assuming S >=Y+2):

        P^2 *(1-P)^Y

and so on, summing each term until your "sliding window" of Y contiguous correct symbols bumps into the end of the block of S symbols. The resulting sum will be the overall probability of success.

This same analysis would have been directly applicable to another interesting amateur problem from years back. If your probability of error in copying a Morse character under exam conditions is a constant P, your errors are all independent (both obviously questionable assumptions) and the FCC rules required you to copy one solid minute out of 5, then what was your overall probability of passing the code test? :-)

Phil

69. Re: [HFSIG:1461] Pulses carrier propagation tests

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 6 Aug 1996 22:36:14 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608070536.WAA18424@servo.qualcomm.com>

Rendered body type: text/plain

Robert,

Very good work -- even if the results aren't as positive as I would have liked. I was especially interested to see you try different pulse durations and modulating waveforms. Since this is inherently a subjective test (i.e., is the signal "audible" or "annoying" to the SSB listener), giving subjective results is reasonable.

I concede that you've shown that under your specific test conditions, FHSS interference can be subjectively annoying to a human listener, even if it doesn't necessarily interfere with communications. This is a useful result. It means we need some form of dynamic coordination with ones' neighbors if SS is to be used operationally on the ham bands under similar conditions. I believe this is a practical idea, and that a flat prohibition of SS in the rules is still not necessary.

Indeed, it's not too hard to conceive of a scheme whereby a local narrowband user broadcasts his listen frequency on a VHF packet channel, alerting a nearby SS transmitter to "cut a hole" in his emission. Or it could be much simpler than that, by simply monitoring for strong local narrowband signals at the SS receiver and temporarily cutting them out of the hopping sequence for some period of time, such as a few minutes. I.e., the narrowband user need only transmit on the channel when the SS station is listening and that would do the trick. (I do confess this wouldn't necessarily work for split-frequency DX operations -- which brings us back to "protected DX subbands".)

Anyway, think of this adaptive channel-blanking scheme as a variation of my MACA multiple access algorithm, which is in turn based on "busy tone multiple access" (BTMA). The common theme to all three is the ability of a receiver to say "shhh, I'm listening" to its neighboring transmitters.

If the total number of hopping channels is large compared to the number of blanked channels, the SS transmitter probably wouldn't even have to tell the receiver about the new hopping sequence. FEC coding would fix the gap at the receiver.

Anyway, there are all sorts of clever ways one might mitigate SS/narrowband interference; I came up with this one in just the few minutes that I've been typing this message. If we succeed, I think the FCC would look quite favorably on us as they've been wrestling with these sorts of dynamic spectrum sharing issues for a long time. The ham bands are a perfect testing ground for this kind of thing. And I can think of other applications for adaptive channel protection -- operating in radio quiet zones, for example. Instead of having to stay off the air entirely within the zone, you'd monitor a special packet channel and forego transmitting only when the protected receiver is actually in operation.

My one major quibble with your results comes from your 30dB extrapolation from the whip RX antenna to a "good amateur station". I would like to see you try it with a *real* "good amateur station". I.e., a local ham friend, if one is close enough to you, especially if he/she has a beam that can be rotated to test the effects of antenna directivity.

In the further work category, it would be interesting to try some objective tests of simulated FHSS interference against existing amateur digital schemes. It would be especially interesting to see how Clover, Pactor, G-TOR etc compare to each other against this kind of pulsed QRM. If the FEC in these schemes are able to handle it well, then one possibility is to permit spread spectrum within those band segments where these schemes are currently popular. Of course, being smaller than a whole band the "hit rate" on any given channel would be greater. But with the increasing popularity of these schemes, perhaps the hopping range will soon be extended to the whole "CW" portion of each band. :-)

Hmm. Perhaps instead of having band segments for "DX" and "local" (or perhaps in addition), we could divide up the HF bands into "machine" and "manual" subbands. Existing CW and SSB/AM would be combined into a "manual" subband, while the "machine" section would accomodate all digital schemes intended for machine reception, including spread spectrum. The thinking, of course, is that emissions in the "machine" subband would be likely FEC coded in a way to resist the burst interference of FHSS while the "manual" emissions would not.

And there is again the possibility of simply not hopping at all unless
you have to (e.g., when the band is evenly crowded). But this leaves
us with another question -- given the clear benefits of FEC coding,
which substantially reduces power requirements at the expense of
bandwidth, how much wider should such signals be allowed to be
assuming they don't take up the whole band?

One approach to the problem is to count how many nearby neighbors N
you are likely to have on the band at any time and let each use no
more than 1/N of the band. That way you can preserve just enough
orthogonality to solve the near-far problem without denying everyone
the benefits of at least some bandwidth expansion. This is really just
an explicit, manual form of the "adaptive notching" mentioned above,
and I tend to prefer automatic methods over manual ones.

Comments?

Phil

70. Re: [HFSIG:1446] Re: Amateur culture

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 7 Aug 1996 01:27:48 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608070827.BAA18788@servo.qualcomm.com>
Rendered body type: text/plain

>As it applies to the current HFSIG discussions, I think that part of
>our educational process should include investigating new techniques and
>technologies. On that basis, I'd love to see some exploration of SS
>techniques on HF. But since nowhere in Part 97 is the rationale for
>Amateur Radio stated as the development of a maximally efficient
>communication system, it's not a given--at least to me--that we should
>abandon all narrowband activity in favor of an SS system even if such
>a system can be shown to be more spectrum-efficient in theory and
>practice. Which means that any SS system must coexist with narrowband
>uses of the spectrum by whatever means make that possible.

There are compelling reasons to investigate spread spectrum that go
beyond spectral efficiency -- as important as that may be, both to us
directly and as a testing ground for other services. SS is
increasingly important as a commercial technology, and this trend is
certain to continue. If the Part 97.1 notion of "training a reservoir
of skilled technicians" is to have any relevance to the outside world,
this is a technology that amateurs should be exposed to, preferably in
as wide a variety of settings (e.g., frequency bands) as
possible. This doesn't mean completely displacing traditional
narrowband operations, but it does mean that reasonable accomodations
should be made for it instead of banishing it entirely to the upper
microwave spectrum (the NIMBY syndrome).

Phil

71. RE: [HFSIG:1454] EVM56002 ?

From: Stephan Sykes <ssykes@emirates.net.ae>
Date: Wed, 7 Aug 1996 06:46:15 +0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB847C.8A8E18C0@csa106.emirates.net.ae>
Rendered body type: text/plain

If it is around the price you quoted, please count me in.

Steve Sykes KD2OM/A61AA

From: Greg Jones[SMTP:wd5ivd@tapr.org]
Sent: Tuesday, August 06, 1996 8:26 AM
To: hfsig@tapr.org
Subject: [HFSIG:1454] EVM56002 ?

There is an opportunity for TAPR to do a group purchase on the Motorola
EVM56002 at a pretty low cost.   Between 20 and 200 units would be the
goal...same price no matter how many are purchased. Figure about $60 under
current commercial prices. No hard prices, becuase more work would have to
be done.

Would there be interest in this in the HF SIG ? Espiecally since Johan's
code would run on it.

I have spoken with Johan and he thinks it would be great...what are some

other reactions.

Cheers - Greg, WD5IVD

72. Re: [HFSIG:1432] Re: Wide band modes

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Tue, 6 Aug 1996 11:08:32 -0400
To: hfsig@tapr.org
Message-ID: <208A69E0.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

>You are exactly right -- different convolutional codes can indeed be
>used to differentiate between signals; it's an entirely viable form
>of CDMA. If the code rate is low enough, then this is virtually
>identical to coded direct sequence spread spectrum! In fact, Andrew
>Viterbi invented such a scheme a few years ago and published a paper
>about it.

Phil,

What is the name of the paper you are talking about here?

Tony

73. Re: [HFSIG:1429] Re: SS on HF bands

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Tue, 6 Aug 1996 11:24:35 -0400
To: hfsig@tapr.org
Message-ID: <208A7380.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

While experimentation and self-training are certainly important
reasons to keep ham radio going, they are no the ONLY reasons. Has
anyone seen the movie 'Independence Day' ? After the aliens had
basicly taken the world over, all satellite communications were
broken. World governments could not communicate with each other.
However, there was one way they could - amateur radio using morse
code! There is a scene where radio operators are franticly using keys
to relay information.

This was a very positive image for ham radio operators. Keeping
amateur radio going for BOTH emergency communications AND advancing
the state of the art is why amateur radio should be kept alive.

Ham radio is NOT just a hobby!!!

73s de
Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1429] Re: SS on HF bands
Author: hfsig@tapr.org at BALT.SMTP
Date: 8/5/96 12:40 AM

I guess it boils down to what you think ham radio is all
about. Reliable communications? That description hardly fits direct
long-haul HF. Easily accessible to those without large lots, liberal
housing covenants and money for big antennas and power amplifiers? I
think not. The popularity of VHF/FM and repeater networks shows there
are a lot of hams who are interested primarily in reliable local
communications with small stations, and who aren't purists when it
comes to using relays. Spread spectrum promises to provide the same
capabilities over wider areas on HF.

I confess that perhaps my view of ham radio is somewhat different than
yours. About the only activity that has a hope of justifying our bands
for very long is technical self-training, experimentation and
advancement. That means keeping up with the state of the art in the
non-amateur world, and spread spectrum is certainly one technology
where we trail the commercial and military world. While I don't really
have anything against DXing and contesting, I really don't see how they
help justify ham radio's existence to the outside world; they're really
little more than technological stamp-collecting.

I don't want to do away with traditional hamming, but I don't want it
to choke experimentation either. While I can't guarantee that there

won't ever be any SS interference to narrowband modes, I think it can
be minimized by careful system design and operational coordination.
And whatever risks there are will be worth the benefits. After all,
ours is not a critical safety-of-life service like aviation; we're
supposed to be experimenters, and experiments can fail. That's part of
the process.

Phil

74. Re: [HFSIG:1464] Re: Amateur culture

From: Pete Hardie <hardie@duke.usask.ca>
Date: Wed, 7 Aug 1996 11:01:43 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.95.960807103956.23754A-100000@duke.usask.ca>
Rendered body type: text/plain

On Tue, 6 Aug 1996, Phil Karn wrote:

> Ragchewing, contesting and DXing seem to be 99.9999% of ham radio.
> While I don't really object to these, it's hard to argue that they
> carry much weight with the non-amateur public that owns our spectrum.
>
> ....
> [They] engage in what can best be described as electronic stamp-collecting.

Two points Phil:

1. Without those 99.9999%, the remaining .0001% would not have *any*
spectrum to experiment in at all.

2. The vast majority of amateurs (myself included) do not have the ability
or, to be generous, sufficient education to ever be in a position to
experiment with SS or other advanced digital modes of communication.

I think the idea that amateurs ever made any significant advances in radio
technology is largely a myth. There's no way that your average ham, 60 or
more years ago, could start with a CW/AM rig and tinker around with it and
come up with SSB. Any ham who did make an advance in technology was, like
Phil, a highly educated person who already had the background to be able
to understand what they were doing *without* a ham radio license. All the
license did for them was give them access to spectrum where they could
experiment.

73 de Pete (still struggling to make that EZ-KIT transmit a packet :-)
ve5va.qrp@usask.ca

75. Re: [HFSIG:1471] Re: Amateur culture

From: Charles Brain <chbrain@dircon.co.uk>
Date: Wed, 07 Aug 1996 22:11:15 +0100
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19960807211115.0067efd0@popmail.dircon.co.uk>
Rendered body type: text/plain

I would like to disagree with the following because if it had not been for
amateur radio I would never have got interested in electronics, never
would have done a degree in the subject and never would have earned
a living doing electronics.

My first introduction to the hobby was at the age of eleven when I used
to listen to old timers Rag chewing on 160 m.

Also an amateur in the early days was reported to have told Marconi
when he met him that he was 'only' an amateur, Marconi replied
so am I.

>I think the idea that amateurs ever made any significant advances in radio
>technology is largely a myth. There's no way that your average ham, 60 or
>more years ago, could start with a CW/AM rig and tinker around with it and
>come up with SSB. Any ham who did make an advance in technology was, like
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>to understand what they were doing *without* a ham radio license. All the
>license did for them was give them access to spectrum where they could
>experiment.
>
>73 de Pete (still struggling to make that EZ-KIT transmit a packet :-)
>ve5va.qrp@usask.ca
>

Regards Charles

76. Re: [HFSIG:1471] Re: Amateur culture

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 7 Aug 1996 18:02:19 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608080102.SAA19732@servo.qualcomm.com>
Rendered body type: text/plain

>1. Without those 99.9999%, the remaining .0001% would not have *any*
>spectrum to experiment in at all.

Why do you say that? The US rules specifically mention
experimentation, training and development. Presumably that (and public
service and international goodwill) is why they gave us spectrum.
They don't mention ragchewing, DX and contesting, one way or the
other.

>2. The vast majority of amateurs (myself included) do not have the ability
>or, to be generous, sufficient education to ever be in a position to
>experiment with SS or other advanced digital modes of communication.

Experimentation can be at many levels. One does not have to build the
technology up from the ground to use it or at least experiment with it
and see how well it works. Airtouch just announced that they're going
to provide Powerband service (their name for Qualcomm CDMA) at the
Republican Convention next week here in San Diego. And if your average
Republican can use spread spectrum, *anyone* can. :-)

>I think the idea that amateurs ever made any significant advances in radio
>technology is largely a myth. There's no way that your average ham, 60 or

Well, it's certainly true that the vast majority of hams have made no
significant advances in radio. But I strongly disagree with your
apparently absolute statement. Yes, many of the advances I attribute
to ham radio were in fact made in the commercial realm by people who
happened to have ham licenses. But even then, ham radio still deserves
at least some of the credit.

In my own case, I got my first ham license at age 14. I credit it with
steering me toward my career. Since graduation, I've managed to
maintain a symbiotic relationship between my ham radio and work
activities. And I'm finding this is fairly common among my colleagues
who are also hams. One of my company's founders, Klein Gilhousen
(WT6G) is fond of saying that "anything you do in ham radio you'll
probably find a use for at work within 6 months." I think I've
demonstrated that several times already. Some examples:

I came up with an improved TCP round trip timing mechanism largely as
a result of my experience in running TCP/IP over amateur packet radio.
It's now a required part of the TCP spec.

I (re)invented what became Bellcore's S/KEY one-time password scheme one
night while visiting a fellow ham in NJ. I wanted to log back into my
UNIX system at home without typing my password in the clear on 2m.

My NOS software has become the foundation of quite a few commercial dialup
IP router packages. Etc.

There are other hams who could list similar accomplishments. Some of
Franklin Antonio's graphics work in Instant Track went into the early
Qualcomm Omnitracs dispatch displays. Quite a few of the RF designers
around here got their first exposure to this stuff through ham radio.
Those who developed Clover, Pactor and the like have seen their
systems used by non-amateur services. AMSAT's pioneering work in small
LEO satellites is now spawning a whole new industry (which,
ironically, is now threatening our spectrum :-(). And so on.

I think it's fair to say that this sort of stuff carries far more
weight with the FCC than does DXing, ragchewing and contesting, even
though only a small fraction of the amateur population is involved in
the former. And that's one reason why the FCC has actually been trying
to promote greater use of spread spectrum in the amateur service.

Phil

77. Re: [HFSIG:1472] Re: Amateur culture

From: lynnc@sos.net
Date: Wed, 7 Aug 1996 20:04:43 -0700
To: hfsig@tapr.org, hfsig@tapr.org

Message-ID: <199608080304.UAA04838@sos.sos.net>
Rendered body type: text/plain

Well, blood boiling now, not a good time to enter feelings, but by gosh, we DID go from CW/AM to SSB and much, much more 60 (well, almost) years ago. Formal training was ARRL license manual, ARRL handbook, and perhaps for the more affluent, RADIO HANDBOOK.

Still in high-school and with cheering older hams around, we did create, innovate and build new and exciting stuff - - daily. We were probably not even average, and probably re-invented most things because technical knowledge didn't spread as fast and seamlessly as it does today. I'm darn proud of my "average" ham friends of the late 1940's and early 1950's--- W7KGV, W7LAN, W7LFA, W7FXD and hoards of others.

I was pretty slow, and didn't come up with my first junk-box SSB rig until 1960. Most of my friends had graduated to commercially built SSB gear by then, and were into home-brew VHF stuff from the high class surplus of the time.

There is a place for everyone in this mess, we had better stick togather more tightly than in the past, as the benevalent public is beginning to look for more space, and the monitoring public would axe us in an instant if they hear the stuff for which we are using our precious bands. (not that the TV and cellular frequencies are a model to admire.)

but I digress------- Lynn, W7LTQ

78. Re: [HFSIG:1457] Re: EVM56002 ?

From: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Date: Wed, 07 Aug 1996 23:21:45 -0400
To: hfsig@tapr.org
Message-ID: <199608080321.XAA10065@appstate-01.campus.mci.net>
Rendered body type: text/plain

At 06:49 AM 8/6/96 -0500, you wrote:
>Greg Jones wrote:
>
>> There is an opportunity for TAPR to do a group purchase on the Motorola
>> EVM56002 at a pretty low cost. Between 20 and 200 units would be the
>> goal...same price no matter how many are purchased. Figure about $60 under
>> current commercial prices. No hard prices, becuase more work would have to
>> be done.
>>
>> Would there be interest in this in the HF SIG ? Espiecally since Johan's
>> code would run on it.
>
>Sounds great considering the price is already only $149.95 at the
>Design-Net price page (http://www2.motorola-dsp.com/dsp/home/net/pg.html).
>
>They also offer a student discount of 20% reducing the price to $120 + $3
>shipping (http://www2.motorola-dsp.com/dsp/home/edu/discount.html)
>
>Mike N4CNW
>
>

If we are talking about less than $100 count me in!

73's
Don, K4JPJ

79. Re: [HFSIG:1469] Re: Wide band modes

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 7 Aug 1996 21:38:15 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608080438.VAA20197@servo.qualcomm.com>
Rendered body type: text/plain

> What is the name of the paper you are talking about here?

"Very Low Rate Convolutional Codes for Maximum Theoretical Performance of Spread-Spectrum Multiple-Access Channels", Andrew J. Viterbi, IEEE Journal on Selected Areas in Communications, Vol 8 No 4, May 1990, p 641.

Abstract - This paper treats a spread-spectrum multiple-access communication system for which both spreading and error control is provided by binary PSK modulation with orthogonal convolutional codes. Performance of spread spectrum multiple access by a large number of users employing this type of coded modulation is determined in the presence of background Gaussian noise. With this approach and coordinated processing at a common receiver, it is shown that the

aggregate data rate of all simultaneous users can approach the Shannon
capacity of the Gaussian noise channel.

80. New RF safety standards

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 8 Aug 1996 01:52:18 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608080852.BAA20871@servo.qualcomm.com>
Rendered body type: text/plain

Some of you may have just heard that the FCC has adopted new
and more stringent rules regarding RF emission safety, and these
rules do NOT exempt amateurs as they have in the past.

See http://www.arrl.org/news/rfsafety for details.

It does occur to me that this might be the impetus the amateur service
needs to finally get serious about using the minimum power required to
maintain communications. It may even spark interest in newer, more
power-efficient modulation methods -- even if they need more power.

And who knows? We might even find our TVI/RFI problems reduced too.

Phil

81. Re: [HFSIG:1473] Re: Amateur culture

From: Mike Curtis <wd6ehr@kaiwan009.kaiwan.com>
Date: Thu, 8 Aug 1996 02:05:57 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608080905.CAA18844@kaiwan009.kaiwan.com>
Rendered body type: text/plain

>
> >1. Without those 99.9999%, the remaining .0001% would not have *any*
> >spectrum to experiment in at all.

> I think it's fair to say that this sort of stuff carries far more
> weight with the FCC than does DXing, ragchewing and contesting, even
> though only a small fraction of the amateur population is involved in
> the former. And that's one reason why the FCC has actually been trying
> to promote greater use of spread spectrum in the amateur service.

One big problem I see with amateur radio is that the cart has been put
before the horse. The "99.9999%" has made it quite difficult for the
innovators to experiment. Yet innovation is a major portion of the
cornerstone of amateur radio. In areas like Los Angeles, it is doggone
nearly impossible to experiment on popular bands such as 2 meters and open
HF bands. 220 is better, but only because laws were changed that MANDATED
setting aside spectrum for "weak signal" stuff. We have old, established
technologies putting ironclad claims on most of the spectrum, and this is
IMHO exacerbated by actually encouraging prospective hams to memorize by
rote the test answers. Yes I know we live in the age of instant
gratification. I just think it's inherently evil.

 -- mike

82. Re: [HFSIG:1448] How much QRM is too much?

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 8 Aug 1996 02:30:47 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608080930.CAA20982@servo.qualcomm.com>
Rendered body type: text/plain

>For someone trying to copy a signal right at the thermal, cosmic, or
>atmospheric noise level, any interference is appreciable. For some hams

"Any" is a rather absolute term; it reminds me of the absolutist
position on biological radiation exposure (no artificial radiation
whatsoever is tolerable even though the natural background can vary
substantially). But unlike that particular can of worms, the math
here is not too hard.

I've been meaning to work out the exact figures for how long an
observing interval is required to even detect a certain small change
in the noise floor with a certain confidence. Tom Clark (W3IWI) does

essentially this kind of calculation all the time in radio astronomy, where all the signals he observes are essentially noise (at least until the first positive SETI acquisition). He sketched out the basic principles for me over lunch a while ago. Once I translated it from "radio astronomerese" into "communicationese" I realized it was all the same math I'm used to seeing in comm theory.

Qualitatively, the minimum required observing interval depends on both the difference in noise level (larger differences can be detected more quickly) and the observation bandwidth (detection is quicker with larger observation bandwidths -- so the obsession with narrowband receivers works to my advantage here! :-))

Phil

83. [HFSIG:1473] Re: Amateur culture

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 8 Aug 96 11:13:03 +0100
To: hfsig@tapr.org
Message-ID: <H000029202318a16@MHS>
Rendered body type: text/plain

Hi Phil,

> I think it's fair to say that this sort of stuff carries far more
> weight with the FCC than does DXing, ragchewing and contesting, even
> though only a small fraction of the amateur population is involved in
> the former.

Yip, I agree, but the 'DXing, ragchewing and contesting' and the likes is where it all starts, and is what keeps it alive in the general population. If amateur radio was only the 0.0001% expert experimenters I would have probably never heard of it, and neither would most of the 0.0001%.

'Technological stamp collecting' is in my opinion an essential part of amateur radio, not to be lightly disregarded.

> And that's one reason why the FCC has actually been trying
> to promote greater use of spread spectrum in the amateur service.

I would hope the FCC is doing this with a wider view of amateur radio, or does it intend to promote the amateur bands as a test bed for commercial experimentation, rather than for more personal radio experiments? Is this how it is justifying an 'amateur' allocation in a commercial enviroment?

This issue caused a bit of controvercy in New Zealand when a number of hams with commercial interests started using ham bands for commercial experimenation, although not blatently. Others saw it as a usefull contribution to amateur radio. I wouldn't like to say, it sure is on the fine line between 'advanceing the art' and 'pecuniary interest'.

Certianly, that line is oversteped when 'pecuniary interest' takes presidence over other legitimate uses of the bands. (thinking more of the FCC here)

Rob

84. Re: [HFSIG:1480] Re: Amateur culture

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 8 Aug 1996 09:07:37 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608081407.JAA22454@www.kelly-afb.org>
Rendered body type: text/plain

Rob,

In your message you write:
[Stuff Deleted]
>
> This issue caused a bit of controvercy in New Zealand when a number of
> hams with commercial interests started using ham bands for commercial
> experimenation, although not blatently. Others saw it as a usefull
> contribution to amateur radio. I wouldn't like to say, it sure is on the
> fine line between 'advanceing the art' and 'pecuniary interest'.
>
> Certianly, that line is oversteped when 'pecuniary interest' takes

> presidence over other legitimate uses of the bands. (thinking more of
> the FCC here)
>
>
> Rob
>
>

        In past years, educational institutions used hambands extensively
        for education/training. Many high schools had ham radio as in part
        of the "shop" program and most electronic technical schools also
        used ham radio as a teaching tool. While I don't think that
        colleges and/or universities used ham radio as a tool at the under-
        graduate level, I know that is was used in lab work for post-
        graduate degrees. And I have to admit that I believe in most cases
        high schools, technical schools and colleges/universities were not
        too concerned whether or not a student had a ham radio license as
        long as there was a ham present in the "lab".

        Also, I know of three specific instances where the FCC has granted
        licenses to commecial businesses to operate in the hambands...granted
        there were frequency restrictions, and special licenses...even
        restrictions on who the stations could talk with...in some instances,
        they were issued special callsigns, other times regular ham callsigns.
        These stations were licensed to do experimental and/or production unit
        field test.

        What's new?

Walt/K5YFW


```
=========================================================================
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development.      |
=========================================================================
|                                 |                                     |
|                                 | The greatest dangers to liberty     |
| Walt DuBose - K5YFW             | lurk in insidious encroachment      |
| E-Mail k5yfw@www.kelly-afb.org  | by men of zeal, well-meaning        |
| Business Telephone: (210)925-6081 | but without understanding.        |
|     Home Telephone: (210)696-3196 |                                   |
|                                 |   - Justice Louis D. Brandeis       |
|                                 |                                     |
=========================================================================
```


85. Re: [HFSIG:1478] Re: Amateur culture

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 8 Aug 1996 09:11:10 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608081411.JAA22482@www.kelly-afb.org>
Rendered body type: text/plain

In your message you write:
[Stuff Deleted]
>
> One big problem I see with amateur radio is that the cart has been put
> before the horse.

        The horse has got to go...the horse is old technology and the
        droppings are causing a mal-odor in our hobby.   -- Walt/K5YFW

86. [HFSIG:1481] Re: Amateur culture

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 8 Aug 96 16:23:14 +0100
To: hfsig@tapr.org, k5yfw@www.kelly-afb.org
Message-ID: <H000029202323446@MHS>
Rendered body type: text/plain

> In past years, educational institutions used hambands extensively
> for education/training. Many high schools had ham radio as in part

Granted, and I support such activities, such as JOTA etc. This is well
in the spirit of amateur radio.

> Also, I know of three specific instances where the FCC has granted
> licenses to commecial businesses to operate in the hambands...granted
> there were frequency restrictions, and special licenses...even

> restrictions on who the stations could talk with...in some instances,
> they were issued special callsigns, other times regular ham
> callsigns. These stations were licensed to do experimental and/or
> production unit field test.

I'm sure there are cases where such commercial use is justified, and I never said I opposed the semi-commercial experimentation in New Zealand (they did not have special commercial licences), I'm just wondering about the fine line between commercialisation of the ham bands and the advancement of technology. International treaties specifically exclude 'pecuniary interest' in amateur radio. I'm wondering if perhaps the FCC are promoting commercial interests *over_and_above* the interests of all amateurs, and possibly against the International Radio Regulations. Perhaps the FCC see the amateur bands as more for experimentation for the advancement of commercial technology, rather than for the self training of amateurs as specified in the international regulations.

Commercial enterprises can still experiment on other bands with other licences, but they cost more money than the ham bands. The non-commercial nature of amateur radio works too ways, to protect industry from amateurs exploiting their allocations commercially, and to protect amateurs from commercial organisations trying to exploit a cheap resource to the detriment of amateurs, and their self training.

I was wonder if this is why they appear to support spread spectrum (an emerging commercial technology) on HF, apparently disregarding the interests of the majority of amateurs.

Rob

87. New floating point EVM from TI

From: forrerj@peak.org (Johan Forrer.)
Date: Thu, 08 Aug 1996 10:18:45 -0800
To: HFSIG@tapr.org
Message-ID: <199608081729.KAA23715@PEAK.ORG>
Rendered body type: text/plain

Hi All,

Those interested in dabbling with DSP hardware; TI's new 320C31 DSK is finally available. Marshall (1-800-522-0084) has them at a special introductory price of $79.

This is a FLOATING POINT DSP chip. I'm not sure about the clock speed, CODEC, or how much memory the DSK has, but I'll let you know when I receive mine.

I have a small collection of software tools for this DSP and am particularly interested in the floating point architecture for implementing FFT demodulators where the extra dynamic range is essential.

Exciting times - there also is some other interesting developments from Motorola that I'll tell you more about in the near future.

--Johan, KC7WW

88. RE: EVM56002 ?

From: rhiii@pop.erols.com (Richard H)
Date: Thu, 08 Aug 1996 18:17:00 GMT
To: hfsig@tapr.org
Message-ID: <320a2920.44799027@smtp.erols.com>
Rendered body type: text/plain

Question to those familiar with this kit.

What is the knowledge entry level for this DSP kit ? Does it assume the student already be fluent in FFT, Z and LaPlace transforms. Can this be considered a "First" DSP learning kit or is it targeted towards the practicing dsp engineer ?

Thanks,
de Richard/N2JR

89. Re: [HFSIG:1480] Re: Amateur culture

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 8 Aug 1996 12:31:19 -0700 (PDT)

To: hfsig@tapr.org
Message-ID: <199608081931.MAA14708@servo.qualcomm.com>
Rendered body type: text/plain

>I would hope the FCC is doing this with a wider view of amateur radio,
>or does it intend to promote the amateur bands as a test bed for
>commercial experimentation, rather than for more personal radio
>experiments? Is this how it is justifying an 'amateur' allocation in a
>commercial enviroment?

Amateur radio has never existed in a vacuum. Although direct
commercial exploitation is not allowed, it has always been one of the
goals of the amateur service to develop technology and to train
engineers and radio operators who can then benefit commercial and
government activities. It's a win-win situation. Not only does that
create new technology for our use, but it earns lots of brownie points
for ham radio.

I believe "commercial exploitation" should be narrowly defined as
selling a public communication service for profit. Developing new
technology on the ham bands that may then be commercialized is in
itself perfectly legitimate, although I am concerned by the recent
trend towards keeping such technology proprietary.

Phil

90. Re: [HFSIG:1482] Re: Amateur culture

From: mwestfal@csci.csusb.edu (Michael Westfall)
Date: Thu, 8 Aug 1996 16:46:35 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608082346.QAA22063@csci.csusb.edu>
Rendered body type: text/plain

> > One big problem I see with amateur radio is that the cart has been put
> > before the horse.
>
>
>   The horse has got to go...the horse is old technology and the
>   droppings are causing a mal-odor in our hobby.   -- Walt/K5YFW

"The warp-drive thrusters have been put before the starship" ???
>
>

--

---
73 de Mike "They call me `MiKUY'", N6KUY
                ax.25net: N6KUY@W6JBT.#SOCA.CA.USA.NOAM
                internet: mwestfal@.csci.csusb.edu
                     web: http://web.csusb.edu/public/csci/mwestfal
Linux: the Gates of Hell shall not prevail.
---

91. Re: [HFSIG:1485] RE: EVM56002 ?

From: forrerj@peak.org (Johan Forrer.)
Date: Thu, 08 Aug 1996 19:35:18 -0800
To: hfsig@tapr.org
Message-ID: <199608090246.TAA26420@PEAK.ORG>
Rendered body type: text/plain

Richard,

>Question to those familiar with this kit.
>
>What is the knowledge entry level for this DSP kit ?   Does it assume
>the student already be fluent in FFT, Z and LaPlace transforms.

Not quite a requirement, but that surely comes in handy. Without that, you
should consider joining a support group where you can share the experiences
of others. In time you will broaden you horizon and the formiddable math

will become a lot of fun.
Since you already know the buzz words, I would say that you are well on your
way.

>Can this be considered a "First" DSP learning kit or is it targeted towards
>the practicing dsp engineer ?
>
>Thanks,

>de Richard/N2JR
>
>

There are of course ways to learn about DSP without having DSP hardware,
i.e., PC program packages. However, if you are in a position where you would
like to use DSP hardware in your experiments or your career track, the EVM's
are often the key to getting started. That goes for engineers or scientific
minds. It does take a combination of some skill, level of interest,
persistence, and intuition to tame the little beasty.

Hope this helps.

--Johan

92. Re: [HFSIG:1487] Re: Amateur culture

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Fri, 9 Aug 1996 07:49:17 -0500 (CDT)
To: hfsig@tapr.org
Cc: karn@qualcomm.com, choffman@pelican.davlin.net (Ric Hoffman),
forrerj@frl.orst.edu (Johan Forrier)
Message-ID: <199608091249.HAA03844@www.kelly-afb.org>
Rendered body type: text/plain

Mike,

In your message you write:
>
> > > One big problem I see with amateur radio is that the cart has been put
> > > before the horse.
> >
> >
> >   The horse has got to go...the horse is old technology and the
> >   droppings are causing a mal-odor in our hobby.   -- Walt/K5YFW
>
>
> "The warp-drive thrusters have been put before the starship" ???
> >
> >

        That's a perfectly acceptable practice as long as you
        SPREAD THEM.   -- Walt/K5YFW

>
>
> --
>
> --
> 73 de Mike "They call me `MiKUY'", N6KUY
>   ax.25net: N6KUY@W6JBT.#SOCA.CA.USA.NOAM
>   internet: mwestfal@.csci.csusb.edu
>   web: http://web.csusb.edu/public/csci/mwestfal
> Linux: the Gates of Hell shall not prevail.
> --
>
>

--
```
=========================================================================
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development.      |
=========================================================================
|                                  |                                    |
|                                  | The greatest dangers to liberty |
| Walt DuBose - K5YFW              | lurk in insidious encroachment  |
| E-Mail k5yfw@www.kelly-afb.org   | by men of zeal, well-meaning    |
| Business Telephone: (210)925-6081 | but without understanding.      |
|     Home Telephone: (210)696-3196 |                                    |
```

```
|                                 |   - Justice Louis D. Brandeis   |
|                                 |                                 |
==========================================================================
```

93. new C31 EVM ?

From: zs6awk@global.co.za (Danie Brynard)
Date: Fri, 09 Aug 1996 17:07:15 +0200
To: hfsig@tapr.org
Message-ID: <199608091904.RAA07706@lin01.global.co.za>
Rendered body type: text/plain

Did anyone play with the new TMS320C31 EVM yet ? How usefull is it for ham radio datacomms experiemnts ? Can one 'flash' applications ? Async Serial Ports ?

Danie
zs6awk@global.co.za

94. ECC for SHORT data block

From: forrerj@peak.org (Johan Forrer.)
Date: Fri, 09 Aug 1996 09:22:46 -0800
To: HFSIG@tapr.org
Message-ID: <199608091633.JAA20720@PEAK.ORG>
Rendered body type: text/plain

Hi Experts,

I wonder what a suitable ECC code would be for a short bit field such as 32-bits.
It need to detect and correct at least six bits. Further, need to have a fast algorithm to compute syndromes and error locations, if it could be done using a table lookup, that would be great.

The field and ECC code is for the header field following the sync vector in my new scheme. I was thinking of using a BCH code.

Any further suggestions?

--Johan, KC7WW

95. FTP mail archive?

From: Robert.Glassey@nmp.nokia.com
Date: Fri, 9 Aug 96 17:52:07 +0100
To: hfsig@tapr.org
Message-ID: <H00002920234821f@MHS>
Rendered body type: text/plain

Hi

I was wondering what has happended to the mail archive on the TAPR FTP site? It seems to have stopped in June.

Cheers,

Rob

96. Re: [HFSIG:1488] RE: EVM56002 ?

From: Richard Harrison <rhiii@pop.erols.com>
Date: Fri, 09 Aug 1996 07:31:20 -0700
To: hfsig@tapr.org
Message-ID: <320B4BB8.6A67@pop.erols.com>
Rendered body type: text/plain

Thank you Johan for taking the time to respond. Taking all of your comments under advisement, will order the EVM56002 and begin the journey towards understanding these new technical building blocks.

I am an EE but it has been quite a while since I've had to work with integro-differential equations and their associated transforms.

Regards,

de Richard/N2JR

97. Free Near-Field Analysis Software

From: k6sti@n2.net (Brian Beezley)
Date: Fri, 9 Aug 1996 12:52:18 -0700
To: cq-contest@tgv.com, moon-net@vm.stlawu.edu, hfsig@tapr.org,
wf1b-rtty@ve7tcp@ampr.org
Message-ID: <199608091952.MAA11031@ravel.n2.net>
Rendered body type: text/plain

As a result of the recent FCC ruling that mandates RF-exposure limits for amateur stations beginning January 1, 1997, I'm making available at no cost a special version of AO Antenna Optimizer software that calculates electric and magnetic near fields. NF.EXE requires a 386 or better, math coprocessor, VGA, and DOS 3.0 or later.

You can download the 245K NF.ZIP file from ftp://n6nd.nosc.mil. You may copy this free software for others as long as no charge is involved and the software is used for amateur purposes only.

After you unzip the file, see READ.ME for more information. Please carefully read the section on accuracy limitations of near-field modeling.

I hope this software helps hams evaluate their stations for compliance with the new FCC rule. The software should be especially useful at high-power contest, DX, and EME stations.

I'm providing this free software without support. The package includes extensive documentation and 92 example antenna files. I hope you'll refrain from calling, writing, or e-mailing questions about downloading or using the software. Thanks!

Brian Beezley, K6STI
k6sti@n2.net

98. FTP Packet Driver Specification

From: forrerj@peak.org (Johan Forrer.)
Date: Mon, 12 Aug 1996 20:17:06 -0800
To: HFSIG@tapr.org
Message-ID: <199608130327.UAA28713@PEAK.ORG>
Rendered body type: text/plain

Hi,

I would be grateful if someone could point me to the location of the "FTP packet driver specification". I have a suspicion that it may be somewhere at Clarkson, but was not able to locate it.

This is to be used to connect a new HF DSP modem a TCP/IP environment. I do have most of the low-level code sorted out, but miss the documentation to understand what is needed to interface to the network OS.

Thanks much.

--Johan, KC7WW

99. SUMMARY- *at least* Y *in a row* probability question

From: Richard Lay <lay@cod.nosc.mil>
Date: Tue, 13 Aug 1996 06:47:15 -0700
Cc: hfsig@tapr.org, lay@cod.nosc.mil
Message-ID: <32108763.7EDA@cod.nosc.mil>
Rendered body type: text/plain

Thanks to all that helped me with my probability question. I know much more about the problem now. Special thanks to Tim Firman, who spent much time on the problem and answering my questions. I also thank Jim Landis, whose solution appears first below. His is the one I implemented both in C (recursive and non-recursive forms) and MATHCAD (non-recursive only). If anyone would like a copy of either of these please let me know. And finally, Ken Butler went at the problem from a different point of view. I have only just received his solution, but it looks interesting, and I post it as it approaches the problem from another prospective. I received many more replies besides these. Thanks to all who responded. You are the ones who make the internet such a valuable resource.

Regards,
Rich Lay

****
Original question:
If the probability of an individual event occurring (heads on a coin, 3 on a die, an error in a received ALE word) is P, then within a sequence

of S events (S coin flips/die rolls/ALE words received), what is the
probability that *at least* Y of these events will occur *in a row* ?
****
Jim Landis solution:
I have a solution that is a little heavy in computation, but it does
solve the problem exactly, with out double counting or miscounting
any cases.

I don't know of any simple solution to that problem in "closed"
notation, but it yields very easily to a recursive formula, i.e. a
formula defined in terms of itself.

Define f(n, Y, S, p) as follows :
f(0, Y, S, p) = 1
f(n, Y,S, p) = 0   for all n > S >= 0
f(n, Y, S, p) = p*f(n-1, Y, S-1,p) + (1-p)*f(Y, Y, S-1, p)

n represents the number of successes needed to complete the current

        "streak", and should initially be set to Y
Y represents the total number of successes needed in a row
S is the number of remaining trials
p is the probability of success on any trial

f() could be interpreted as "the probability that you will have a
streak of n successes immediately, or a streak of Y successes later
on". The two terms evaluate the remaining probability after each roll
of the die. First term corresponds to success on the current roll, the
second term to failure.

So the probability of an unbroken string of four 3's in 20 rolls of a
fair die is f(4,4,20, 1/6)   or approximately 0.011026 .

You can verify that Y=1, S=1 yields a probability p, as you would
expect. You can also verify that Y=S yields p^^S.

This can be implemented in a simple computer program. However, because

of the double recursion, the run time of the program is O(2^^(S-Y)).
The formula is essentially called once for each of the 2 to the power
of S outcomes. For S=20, that is more than a million times. Beyond
S = 30 or so, that approach is so slow as to be impractical.

A non-recursive program could be written based on an S +1 by Y+1
array of values, and running in time O((S*Y)^^2). The approach would
be to create an array. Initialize one column of the array to value 1,
based on the first rule. Initialize one half of the array to 0, as
per the second rule. Every remaining cell of the array can be
calculated from 2 other cells by the function definition.
****
Ken Butler solution:
Anyway, assuming that independence and constant probability are OK here,
my
line of thinking is the following: let N be the total number of words
sent
when Y consecutive words are received incorrectly for the first time. For
example, if Y=3, in the following sequence of words sent (C = correctly
received, I = incorrect):

I C I I C C I I I

N = 9, since 9 words have been sent by the time the 3rd consecutive I
appears. Relating this to your problem, if N>S, then the radios link up,
whereas if N <= S, the link is disestablished. So prob(N>S) is the
probability you want. (This takes care of the "at least" problem, because
once Y consecutive I's show up, we stop looking.) You also see that the
last
four (=Y+1) words sent are a C followed by K I's; this must be the way
the
process stops, because if we'd gotten any more consecutive I's, we would
have
stopped earlier, and if we'd gotten any fewer, we'd keep looking.)

To show what happens with Y=2:

N cannot be 0 or 1;
N = 2 if we get I I, whose probability is P^2.
N = 3 if we get C I I, with prob. (1-P)*P^2.

N = 4 if we get I C I I or C C I I, ie. the last 3 are C I I no matter
what
the first is. Probability is (1-P)*P^2 as for N=3.

Then, a pattern:

N = 5 if we survive the first 2 words and then get C I I;
N = 6 if   3   C I I;
..
..
N = m if   m-3   C I I,

so that prob(N = m) = prob(N > m-3) * (1-P) * P^2.

Overall, the probability is 0 for a bit, P^2 once, (1-P)*P^2 for a bit,
and
then the above relationship holds. This was for Y=2, but the same idea
generalizes to other Y:

Prob(N = m) = : 0 if m < Y-1
                 P^Y if m = Y
                 (1-P) * P^Y * prob(N > m-Y-1) if m > Y.

Because each probability depends on previous ones, you'll need to
calculate
these probabilities one at a time, starting from 0 (or Y) and continuing
up
to S. It's probably easiest to keep track of prob(N > m), since this is
the
quantity of interest anyway, noting that prob(N > m) = prob(N > m-1) -
prob(N = m), and prob(N > 0) = 0.
****

100. New modem

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 13 Aug 96 17:58:42 +0100
To: hfsig@tapr.org
Message-ID: <H00002920238df82@MHS>
Rendered body type: text/plain

> This is to be used to connect a new HF DSP modem a TCP/IP environment.
> I do have most of the low-level code sorted out, but miss the
> documentation to understand what is needed to interface to the network
> OS.

Hi Johan,

I wonder how you are getting on with your new modem. Perhaps you would
like to post a brief description of it to the group.

I gather it uses two independent tones with PSK, or is this a multi tone
OFDM or MFSK modem?

Cheers,

Rob

101. Re: [HFSIG:1497] New modem

From: forrerj@peak.org (Johan Forrer.)
Date: Tue, 13 Aug 1996 14:00:50 -0800
To: hfsig@tapr.org
Message-ID: <199608132111.OAA00166@PEAK.ORG>
Rendered body type: text/plain

Hi Rob,

>I wonder how you are getting on with your new modem. Perhaps you would
>like to post a brief description of it to the group.
>
>I gather it uses two independent tones with PSK, or is this a multi tone
>OFDM or MFSK modem?
>
>Cheers,
>
>Rob
>

>

Sorry that I have not posted more details - just not had much time to work on and refine a specification as yet. Hopefully more at the DCC.

A short(?) description for those that cannot wait :-)

As things stand at the moment, it is FDM (not OFDM);four RC-shaped DQPSK waveforms in parallel each at a 75 baud symbol rate - 600 Hz at -60 dB. Modulation is DQPSK that varies from BPSK to 16-DQPSK, selectable to match conditions. The full bandwidth is always used, even at low data rates, however, both time and frequency diversity is then employed. The idea is to use a common DQPSK demodulator that does'nt know the difference - it just demodulates the input waveform as if it were m-DQPSK. That is the job for the packet dissasembler that knows how to unpack the data from information provided in a data header. For that purpose, each packet contains a header that provides a) frequency tracking information, b) bit phasing/clock information (extracted from the sync vector), and c) packet formatting/status information (an ECC protected field). Actual data contents are convolutionally coded using a variable rate code on the inner level, with an outer level block code.

Three modes of operation is planned:

Mode A: Unconnected broadcast mode with full ECC.
Mode B: Hybrid ARQ.
Mode C: Networked (CSMA) with encapsulated TCP/IP.
Mode D: Read-along mode. It is possible for a third party to decode and
        read traffic, pretty much as we can do with packet at the moment.

All specifications will be published for the benefit of those that wish to implement the protocol. It will be an "open architecture" although I hope to encourage commercialization - consulting and user-support services can be arranged. One essential item that would be made available is a "golden" CD-ROM - this will contain the printed standard, also test and diagnostic signals that will help one to verify whether your implementation conforms to the standard or not.

Hope this helps in the interim,

--Johan, KC7WW

102. Sound Cards

From: Mike Kerry <100577.1452@CompuServe.COM>
Date: 14 Aug 96 09:23:49 EDT
To: "(unknown)" <HFSIG@tapr.org>
Message-ID: <960814132349_100577.1452_GHW94-1@CompuServe.COM>
Rendered body type: text/plain

Can anyone please recommend a good PC sound card
which also can be programmed for DSP applications?

(I think this means having the 21xx chip).

Thanks

Mike G4BMK

103. Re: [HFSIG:1499] Sound Cards

From: Johan Forre <forrerj@peak.org>
Date: Wed, 14 Aug 1996 08:47:00 -0700 (PDT)
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <Pine.SUN.3.91.960814084143.29776C-100000@kira>
Rendered body type: text/plain

Hi Mike,

There used to be a number of manufacturers that used an Analog Devices three-chip set (included the 2115 DSP) on their sound cards. Now they are very scarce. Look at the Echo Speech DSP I on their web site: www.echospeech.com I also have just gotten a catalog from Jameco and see the sell the Orchid Soundwave 32 for 129 bucks. That is a 20 Mhz version DSP and I have one that I'm very happy with.

The other alternative is to look for a Turtle Beach Tahiti card - that one though has a 56002. There is a software toolkit available for that one as well. It's a really nice PC plugin card.

Hope this helps.

--Johan

On Wed, 14 Aug 1996, Mike Kerry wrote:

> Can anyone please recommend a good PC sound card
> which also can be programmed for DSP applications?
>
> (I think this means having the 21xx chip).
>
> Thanks
>
> Mike G4BMK
>
>

104. Re: [HFSIG:1491] ECC for SHORT data block

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 14 Aug 1996 11:02:28 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608141802.LAA13843@servo.qualcomm.com>
Rendered body type: text/plain

>I wonder what a suitable ECC code would be for a short bit field such as
>32-bits.

Is that 32 data bits plus however much overhead is necessary?

Why use such a small block? Performance is always better for larger
blocks.

The more I think about the problems of a fading channel, the more I am
troubled by schemes that separately encode sync, header and
data. There's something to be said for a scheme that encodes and
interleaves a continuous bit stream and then layers a simple packet
framing scheme on top that doesn't have to deal with frequent errors
(e.g., HDLC). Then the packet header would enjoy as much protection
as a much larger data field.

One way is a convolutional interleaver followed by a convolutional
encoder. A Reed-Solomon code could be layered on top of this to
further steepen the waterfall curve if a sync word is inserted to mark
the beginning of each RS block. Viterbi decoder and deinterleaver sync
can be achieved by a brute-force search of the possible states looking
for a good decoder metric to pop out. This is practical because there
aren't that many states to search. This is similar to the framing
approach used in digital video broadcasting.

The only drawback I can see is the getting-started overhead on short
transmissions. But if you need a certain amount of time diversity on
a fading channel to span the required number of channel coherence
times, you don't really have a choice -- a short transmission could
get swallowed by a fade. I.e., there's a minimum transmission time
requirement.

Phil

105. [HFSIG:1486] Re: Amateur culture

From: Robert.Glassey@nmp.nokia.com
Date: Wed, 14 Aug 96 19:35:28 +0100
To: hfsig@tapr.org
Message-ID: <H000029202338282@MHS>
Rendered body type: text/plain

>>I would hope the FCC is doing this with a wider view of amateur radio,
>>or does it intend to promote the amateur bands as a test bed for
>>commercial experimentation, rather than for more personal radio
>>experiments? Is this how it is justifying an 'amateur' allocation in a
>>commercial enviroment?
>
> Amateur radio has never existed in a vacuum. Although direct
> commercial exploitation is not allowed, it has always been one of the
> goals of the amateur service to develop technology and to train
> engineers and radio operators who can then benefit commercial and
> government activities. It's a win-win situation. Not only does that
> create new technology for our use, but it earns lots of brownie points

> for ham radio.

It's wider than that but yes, thats part of it.

> Developing new technology on the ham bands that may then be
> commercialized is in itself perfectly legitimate

Mostly, but it can get a little blured when the amateurs concerned already have a commercial application and simply use amateur radio as a vehicle to do commercial research. It's better when amateur research finds an application in the commercial world, where the amateurs concerned have no direct commercial interest, but it's very fuzzy.

> I believe "commercial exploitation" should be narrowly defined as
> selling a public communication service for profit.

The international radio regulations defines the amateur service in article S1.56 as:

"A radiocommunication service for the purpose of self training, intercommunication and technical investigations carried out by amateurs, that is, by duly authorised persons interested in radio technique soley with a personal aim and without pecuniary interest."

This definition is not on the adgenda of any upcomming WRC.

Nowhere in the regulations does it mention advancement of the state of the art. The only place it mentions the state of the art is when it requires that 'emmited frequency shall be as stable and free from spurious emmisions as the state of technical development for such stations permits'. Advanceing the art appears to be an addition of the FCC, which of course has no duristiction outside the USA.

Still, advancing the art is one motive for interest in radio technique (and the purpose of HFSIG) and should be encouraged, although never to the detriment of other equaly valid reasons for interest in radio technique. Of course it is usually entirly in the interests of amateurs in general, but the choice must be theirs, it should never be forced upon them especially in the name of some claimed benifit to a few.

Strictly speaking even the likes of CLOVER and PACTOR should have been developed with commercial expreimental licences, then the final tested products sold to amateurs. But this is just as tricky, the line is definatly blured. It's made for dificult when PACTOR II for example is sold to non-amateurs for high prices as a propitary commercial protocol.

The difficulity of course, is the risk of loosing the amateur element of amateur radio to commercial interests, and thereby either effectivly or actually loosing amateur bands, or even the entire service, as it may be seen as simply a commercial service and sold accordingly, with many eager buyers, and an equally eager seller.

Cheers,

Rob

106. Re: [HFSIG:1501] Re: ECC for SHORT data block

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 14 Aug 1996 15:05:55 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608142205.PAA17918@servo.qualcomm.com>
Rendered body type: text/plain

While waiting for a doctor's appt, I sketched out an approach to an extremely fade-resistant coding scheme along the lines I discussed earlier today.

The channel modulation can be any noncoherent scheme, such as DBPSK or M-ary FSK. The channel symbol stream is fed from a convolutional interleaver. Feeding the convolutional interleaver is a convolutional encoder of some rate, 1/N.

The interleaving degree is one more than some integer multiple of N, chosen to be large enough to give an adequate interleaving span for the symbol rate and channel coherence time. The extra input to the interleaver multiplexer is fed with a PN generator used for acquisition and synchronization.

The PN generator period corresponds to one "block length", and the

PN epoch is aligned so that it marks the beginning of the block. The
PN generator runs continuously throughout a transmission.

The FEC "block" could correspond to a fixed-size block of a
sequentially decoded convolutional code, or it could correspond to a
Reed-Solomon block that's layered on top of a Viterbi-decoded
convolutional code operating in stream mode, possibly with an extra
layer of interleaving between the RS and Viterbi coders to spread out
the error bursts from the Viterbi decoder. (This is a standard feature
of such concatenated codes.)

To acquire this signal, you DBPSK (or M-ary FSK) demodulate the
incoming symbol stream at all possible symbol timing phases and then
run a PN correlator with sampling taps spaced according to the
periodic appearance of the known PN sync sequence in the modulated
stream. When the correlator reports a sufficiently high peak, you now
have modem symbol timing, Viterbi decoder phase and Reed-Solomon block
phase. Or in the case of sequential decoding, you know where to begin
sequential decoding.

Alternatively, the "sync channel" input to the interleaver could be
all 1's, and the multiplexed output could be PN scrambled. The
advantage of this approach is that it is obviously quite easily
extended to PN scrambling at a faster rate, i.e., direct sequence
spread spectrum.

The result of all this effort is a series of decoded fixed-size data
blocks. Each block is almost certain to be either perfect or erased,
i.e., there should be few if any undetected errors at this level. Now
you can layer a conventional framing scheme, e.g., bit-synchronous or
octet-framed HDLC and carry a series of variable length packets.

This scheme seems to provide the maximum robustness possible against
fading. Because packets are all framed at a higher layer, no part of the
channel stream is more vulnerable than any other.

The biggest drawback is the fairly high overhead required to fill the
interleaver buffers at the start of transmission and to flush them at
the end, especially if a high degree of interleaving is required (as
it would be when the the symbol rate is relatively high and the
channel coherence time is relatively long). But some overhead is
inevitable if you really want protection against fading. (Consider a
standalone packet small enough to get lost in a relatively slow fade.)
Some tradeoffs are probably required here, as at some point you have
to fall back on retransmission (as when fading is very slow).

Phil

107. Re: [HFSIG:1501] Re: ECC for SHORT data block

From: forrerj@peak.org (Johan Forrer.)
Date: Wed, 14 Aug 1996 21:50:48 -0800
To: hfsig@tapr.org
Message-ID: <199608150501.WAA00823@PEAK.ORG>
Rendered body type: text/plain

Phil,

Thanks for the feedback. You make some good points here and your follow-up
is even more interesting and noteworthy. I'll comment on that shortly.

>>I wonder what a suitable ECC code would be for a short bit field such as
>>32-bits.
>
>Is that 32 data bits plus however much overhead is necessary?

Right, 32 bits data + x coding bits; whatever it takes.

>
>Why use such a small block? Performance is always better for larger
>blocks.

Granted; The header is as important as correct synchronization. Without
that, the message cannot be run through the Fano decoder as one would'nt
know how the bit stream is to be composed. Typically, the block will be a
very compact field.

>
>The more I think about the problems of a fading channel, the more I am
>troubled by schemes that separately encode sync, header and
>data. There's something to be said for a scheme that encodes and
>interleaves a continuous bit stream and then layers a simple packet
>framing scheme on top that doesn't have to deal with frequent errors
>(e.g., HDLC). Then the packet header would enjoy as much protection
>as a much larger data field.

This is a concern; With flat fading, or certain kinds of selective fading,
this certainly may be valid. The sync vector/header, as presented, does not
really exploit time diversity to address fading. Except (and that is part of
the question I posed), if the right kind of code is used, i.e., one that
effectively deals with burst errors, like for example RS.

The proposed scheme does however, offer some strenghts to this problem. It
does exploit a certain degree of frequecy diversity in addition to robust
modulation waveform for this portion; each of the four channels carries the
same information (replicated x 4) in four BPSK identical channels. These two
factors, together, offers some advantages that are put to good use to combat
the HF channel. If I may illustrate the expected gain in error performance
that this robust header may offer:

                For a BER of 10-5 in a 3 kHz BW channel:
1200 bps with 2 fading channels, 2ms multipath, and 1 Hz fading BW req. 11
dB S/N
  75 bps with 2 fading channels, 5ms multipath, and 5 Hz fading BW req. 2
dB S/N
                                                                      ^^^^^

(from MIL-STD-188 documents - some liberties taken)

We are looking at the 75 bps performance figure for the sync vector and
header portion. Think we can agree that this illustrates a fairly realistic,
though poor HF channel, and that the encoding scheme/waveform is not doing
too bad, i.e. 2dB S/N for a BER of 10-5.

--Johan

108. Re: [HFSIG:1504] Re: ECC for SHORT data block

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 15 Aug 1996 03:15:23 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608151015.DAA19279@servo.qualcomm.com>
Rendered body type: text/plain

Some further reflections on the scheme I sketched out earlier today.
My original reason for using a convolutional interleaver was to make it
practical to try all possible interleaver phases in the acquistion phase.

But that's not necessary if I mux in a continuous PN sync
sequence. Just make the PN period equal to the length of a block, and
phase it accordingly. Then when you find the peak from the PN
correlator at the receiver, you know exactly where all the symbols are
in the block and where the block begins and ends.

This gets rid of all overhead from filling and flushing the
interleaver pipeline, since a block interleaver produces exactly as
many symbols as it gets. But it does mean using a fixed size block,
and there may be some internal fragmentation (fancy term for "unused
space") on small packets. But again, this may be inevitable if you're
trying for enough time diversity to overcome fading.

Phil

109. TAPR offers Motorola EVM56002 group purchase

From: Greg Jones <wd5ivd@tapr.org>
Date: Fri, 16 Aug 1996 12:22:26 -0500 (CDT)
To: tapr-bb@tapr.org (TAPR-BB mailing),
        amsat-bb@amsat.org (AMSAT BB Mail Group),
        hfsig@tapr.org (HF SIG mailing), dsp-93@tapr.org (DSP-93 Build)
Message-ID: <199608161722.MAA09958@tapr.org>
Rendered body type: text/plain

TAPR offers Motorola EVM56002 group purchase   8/15/96

TAPR and Motorola have agreed on a special group purchase of the Motorola
DSP56002 Evaluation Modules (EVM) for a limited time. The DSP56002 EVM is a

general purpose DSP board with on board peripherals and software for assembling and debugging code. TAPR would like to thank Tim Baggett, AA5DF, of Motorola for facilitating this purchase.

The price will be:
   $85.00 US (+ s/h) **

         The EVM56002 normally retails through distributors for $150.

         ** No 10% Discount for TAPR Members.
            Due to the nature of the purchase, TAPR cannot offer a
            membership discount.

For more information on the DSP56002EVM purchase, please see the TAPR web page at http://www.tapr.org

For more information on the DSP56002 EVM and Motorola DSP products visit the Motorola DSP web page at http://www.motorola-dsp.com.

Please note: This unit provides a minimum interface for radio connections. Additional circuits are required for specific radio interfaces based on the application. See the TAPR HF SIG Upload Area (ftp.tapr.org /tapr/SIG/hfsig/upload) for files on one type of radio interface. There are other units that provide complete radio interfaces and should be considered if you feel that you are not capable of building such items.

This EVM group purchase will include:

     * EVM56002 Board
        See below for specs.

     * Motorola's DSP56000 cross assembler
     * Domain Technologies' debug GUI debug software (DOS, >386 CPU)
        both communicates with the EVM over an RS232 serial port

     * Available amateur software developed for the 56002 EVM board

     * Documentation
          56000 Family Manual,
          56002 User's Manual,
          56002 Data Sheet,
          Quickstart Guide,
          Debugger manual, and
          Schematics

TAPR will be taking orders for up to 200 units.

Orders will be taken through the ARRL/TAPR Digital Communications Conference in Seattle on September 21-22, 1996.

Shipping will not begin until the units are delivered from Motorola, which will be sometime after September 22nd. Money will be deposited when the order is placed with Motorola.

If you would like to order one of these units, you can
      e-mail tapr@tapr.org,
      phone (817) 383-0000, or
      fax (817) 566-2544.

      Office Hours: Tuesday - Friday 9am - 12pm, 3pm - 5pm Central Time

Questions concerning the unit and details on the buy will be handled on the TAPR DSP-93 Special Interest Group list. To subscribe, send e-mail to listserv@tapr.org. In the message type subscribe dsp93 YourFirstName YourLastName. The server will then send you a message back. Announcements on the status of the shipment will be made to the TAPR DSP-93 list and TAPR-BB lists.

Shipping and Handling
----
* Shipping and Handling within the US will be $10.00 US

by UPS Ground unless otherwise requested by purchaser.

* International Shipping will need to contact the TAPR office and get a quote on the shipping to your country. TAPR uses International Express Mail, unless the purchaser requires something else.

General Information

The DSP56002EVM is based on the 56002 general purpose DSP by Motorola. The 56002 is a 24-bit precision, dual memory space harvard architecture digital signal processor. The board contains a stereo CD-quality audio codec, on-board 32k words external SRAM, and an expansion spot for a 32K flash EEPROM. The user only needs to supply the IBM PC compatible computer, a 7-9 volt AC or DC power supply, and an RS232 cable for interfacing to the PC.

The DSP56002 EVM is perfect for the hobbyist and student wishing to learn DSP programming on the Motorola 56002 DSP. Many amateur radio applications have been written and used on the DSP56002 EVM including, 1200 AFSK packet, 9600 FSK packet, RTTY, BPSK, QPSK, and various audio filters for CW, SSB, and QRM/QRN reduction.

The 24-bit DSP56002 feature:
- 40 MHz operation (new EVM's are 6 MHZ, and most have operated at much higher clock rates)
- 20 MIPS at 40 MHz (50 nS instruction cycle at 40 MHz)
- 120 Million Operations per second (MOPS) at 40 MHz
- 512x24 bit on-chip program RAM with bootstrap ROM
- two 256x24 bit on-chip data RAMs (X and Y data spaces)
- External memory expansion with 16 bit address and 24 bit data bus
- Byte wide host interface

- Synchronous Serial Interface (SSI)
- Serial Communications Interface (SCI)
- 24-bit timer/event counter
- software programmable internal clock PLL

The EVM features:
- 32kx24 bit zero wait-state external SRAM
- option for 32Kx8 flash EEPROM for program bootstrapping and stand-alone operation
- Crystal Semiconductor's CS4215 stereo CD quality sigma-delta A/D D/A
- on-board voltage regulator
- RS232 interface to the OnCE (tm) controller
- RS232 interface to the DSP SCI port for communication between the DSP56002 and a terminal (PC) directly
- connectors for external access to the DSP56002's memory expansion, host interface, and serial interfaces (SSI, SCI)
- documentation (56000 Family Manual, 56002 User's Manual, 56002 data sheet, quickstart guide, debugger manual, schematics)

The on-board Crystal A/D D/A features:
- 24.576 MHz crystal for sampling at 48, 32, 16, 9.6, and 8 Ksps
- 16 bit linear, 8-bit mu-law, 8-bit a-law, and 8-bit linear sampling

- the 24.576 MHz crystal can be replaced to provide other sample rates
- Software programmable signal attenuation

P.S. TAPR is also doing a group purchase on PC-DSP and PC-SIM for Windows. This is an excellent package for learning and developing DSP software. See: http://www.tapr.org

The PC-DSP is an interactive, menu-driven software package used for: waveform synthesis using a variety of methods, basic signal operation, fast Fourier transforms, convolution and correlation, solution of difference equations, analysis and design of IIR and FIR filters, digital filter simulation and code generation, and power spectrum estimation using classical and modern techniques. Some key features of PC-DSP listed include: GNUPLOT support, code generation, macro compiler, dialog compiler, sound file support, data file formats, and compatibility with PC-SIM.

Tucson Amateur Packet Radio
8987-309 E Tanque Verde Rd #337 * Tucson, Az * 85749-9399 * 817-383-0000
---
e-mail: TAPR@TAPR.ORG ftp: ftp.tapr.org web: http://www.tapr.org/tapr
---

110. [HFSIG:1500] Re: Sound Cards -Reply

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Tue, 20 Aug 1996 07:42:39 +0200
To: hfsig@tapr.org
Message-ID: <s2196c2b.055@KIDD.CO.ZA>
Rendered body type: text/plain

Johan commented:
The other alternative is to look for a Turtle Beach Tahiti card - that one though has a 56002. There is a software toolkit available for that one as well. It's a really nice PC plugin card.

Hope this helps.

--Johan

How much is this card more or less Johan ? Does it have a debugger ?
danie

111. Re: [HFSIG:1507] Re: Sound Cards -Reply

From: forrerj@peak.org (Johan Forrer.)
Date: Tue, 20 Aug 1996 08:47:32 -0800
To: hfsig@tapr.org
Message-ID: <199608201556.IAA16578@PEAK.ORG>
Rendered body type: text/plain

Danie,

>Johan commented:
>The other alternative is to look for a Turtle Beach Tahiti card - that one
>though has a 56002. There is a software toolkit available for that one as
>well. It's a really nice PC plugin card.
>
>Hope this helps.
>
>--Johan
>
>How much is this card more or less Johan ? Does it have a debugger ?
>danie
>
>

Don't know the price, but to be competitive, it would be in the $250 range. And yes; there is a 3-rd party software toolkit (not from Turtle Beach) with a debugger that apparently has the usual breakpoint/single step debugging.

While on the subject: if anyone would like to tackle the DSP chip on the Creative Labs. "Soundblaster" boards, i.e., the chip they call the "ASP", I have found the manual for it on the Web. Unfortunately it is a rather limited DSP - better to be left alone (my opinion). Anyway, CL have lately been using 56002's in their video conferencing products.

There also is a TI320C32-based sound card that may interest you; see www.core-dynamics.com for further details. Not sure what the deal is regarding your own software development for it.

A bit of feedback on the TI320C31 floating point DSP that may interest you: That turned out to be a somewhat of a dissapointment for me; the multiplier takes two 32-bit integers and the result is a 40-bit number. Compare this with the 56002 EVM that takes two 24-bit (or 48-bit doubles) but gives you 56-bit accumulator arithmetic results. Or the ADSP 21xx family that takes two 16-bit data words and results are 40-bit numbers. It is possible of course to achieve extended precision through multiple instructions, albeit at a performance hit. Unless one really needs 32-bit floating point multiplies, I think you will agree that the 56002 EVM is an exceptional bit of hardware.

Sorry - nothing to do with HF digital, but hope some useful information for experimentors.

--Johan

112. Re: [HFSIG:1495] FTP Packet Driver Specification

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 20 Aug 1996 19:37:04 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608210237.TAA17725@servo.qualcomm.com>
Rendered body type: text/plain

Johan,

For the FTP Packet Driver spec, see

http://www.crynwr.com/crynwr/packet_driver.html

Note that all this really does is to provide a portable interface to
an Ethernet card. Although the spec is supposedly general enough to
support other kinds of interfaces, I don't know that it has ever been
used for this purpose.

The software that calls a packet driver must still know quite a bit
about the Ethernet link layer, such as addresses, type fields, and the
like. So unless you are prepared to make you modem look just like an
Ethernet board (which is a questionable thing to do for an HF radio
modem) you don't really want to use this interface. What you really
want is a full-blown IP router that can encapsulate and decapsulate IP
in various link layer protocols such as AX25, Ethernet and whatever
link protocol you create for your modem. That would be much more
useful, I think.

Phil

113. Re: [HFSIG:1508] Re: Sound Cards -Reply

From: rickc@tridelta.com (Rick Contaldo)
Date: Wed, 21 Aug 1996 09:01:18 -0400 (EDT)
To: hfsig@tapr.org, hfsig@tapr.org
Message-ID: <199608211301.JAA01480@sparcy.tridelta.com>
Rendered body type: text/plain

At 11:01 AM 8/20/96 -0500, Johan Forrer. wrote:

>While on the subject: if anyone would like to tackle the DSP chip on the
>Creative Labs. "Soundblaster" boards, i.e., the chip they call the "ASP", I
>have found the manual for it on the Web. Unfortunately it is a rather
>limited DSP - better to be left alone (my opinion). Anyway, CL have lately
>been using 56002's in their video conferencing products.
>
>
>--Johan
>
>

Johan,

I'm curios about the Creative Labs ASP. Would you please point me to
the site where you found the manual?

Thanks,

Rick

114. Frame-Check Sequence for AX.25

From: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Date: Wed, 21 Aug 1996 14:12:34 -0400
To: hfsig@tapr.org
Message-ID: <199608211812.OAA04139@appstate-01.campus.mci.net>
Rendered body type: text/plain

I'm trying to code sending packet info and for the Frame Check Sequence
(FCS) I find the AX.25 specs only say, "It shall be calculated in accordance
with ISO 3309 (HDLC) Recommendatons." Anyone know where I could find a
description for the FCS calculation (or even better Motorola 68HC11 code!).

Don, K4JPJ

115. Re: [HFSIG:1510] Re: Sound Cards -Reply

From: Johan Forre <forrerj@peak.org>
Date: Wed, 21 Aug 1996 12:28:00 -0700 (PDT)
To: Rick Contaldo <rickc@tridelta.com>
Cc: hfsig@tapr.org

Message-ID: <Pine.SUN.3.91.960821122524.23532A-100000@kira>
Rendered body type: text/plain

Rick,

On Wed, 21 Aug 1996, Rick Contaldo wrote:

> At 11:01 AM 8/20/96 -0500, Johan Forrer. wrote:
>
> >While on the subject: if anyone would like to tackle the DSP chip on the
> >Creative Labs. "Soundblaster" boards, i.e., the chip they call the "ASP", I
> >have found the manual for it on the Web. Unfortunately it is a rather
> >limited DSP - better to be left alone (my opinion). Anyway, CL have lately
> >been using 56002's in their video conferencing products.
> >
> >
> >--Johan
> >
> >
>
> Johan,
>
> I'm curios about the Creative Labs ASP. Would you please point me to
> the site where you found the manual?

Look for the D-950 core under chips at SGS-Thompson's web site (www.st.com).

>
> Thanks,
>
> Rick
>
>
>

Good luck.

--Johan

116. Re: [HFSIG:1509] Re: FTP Packet Driver Specification

From: forrerj@peak.org (Johan Forrer.)
Date: Wed, 21 Aug 1996 19:38:49 -0800
To: hfsig@tapr.org
Message-ID: <199608220248.TAA04371@PEAK.ORG>
Rendered body type: text/plain

Phil,

Thanks for the pointer and also thanks to all the others that pointed me in
the right direction.

With Pawel's help, I hacked a version of his AX25 low level driver and then
have NOS call the driver through a software interrupt via standard FTP
packet driver call mechanism. Looked simple enough (Heck, I learned a lot.)

I have been testing and playing with this software interface for monitoring
300 baud HF packet, although I have no intentions to use 300 baud HF packet
- just thought it a good test. NOS is used to monitor AX25 traffic, but I
could have used a number of user interface programs that would accept the
interrupt hook.
It is encouraging to see how well the low-level HDLC code seems to do,
considering the poor conditions and very low signal levels on the bands.
However, I must admit that the DSP modem is probably the main contributing
factor.

Anyway, still a lot more to do and a lot more to learn.

73's

--Johan

117. PSK Sound files

From: forrerj@peak.org (Johan Forrer.)
Date: Fri, 22 Aug 1997 10:46:48 -0800
To: HFSIG@tapr.org
Message-ID: <199608221755.KAA20435@PEAK.ORG>
Rendered body type: text/plain

Hi folks,

I uploaded some .WAV sound files from my experimental DQPSK
modems. These will give you an idea what the waveforms sound
and look like. Included are single and multi-channel m-DPSK
samples. The package is available for downloading from:

ftp://ftp.tapr.org/SIG/hfsig/upload/psk_snd1.txt and psk_snd1.zip

Wish I had more time to work on this project, but things have
been busy as usual.

73's

--Johan

118. PSK Sound files

From: forrerj@peak.org (Johan Forrer.)
Date: Fri, 22 Aug 1997 23:49:43 -0800
To: hfsig@tapr.org
Message-ID: <199608230658.XAA05524@PEAK.ORG>
Rendered body type: text/plain

Hi,

If you are having problems locating the PSK sound files, either
try:

http://www.tapr.org using a web browser - follow the links to hfsig/upload, or;

FTP directly to: ftp.tapr.org/tapr/SIG/hfsig/psk_snd1.txt + .zip

That should do it. Sorry about that.

--Johan

119. New member

From: vk7zo@calvados.apana.org.au (Graham Ranft)
Date: Fri, 23 Aug 96 09:07:22 +1000
To: hfsig@tapr.org
Message-ID: <Za39uD2w165w@calvados.apana.org.au>
Rendered body type: text/plain

Hi just a short one to say hello and introduce myself on HFSIG.
Recently aquired a paccom ptc level 1 and this has started me looking
again at digital sig precessing and digging into coding etc as well as
things ionospherical... I am currently only operation on 3632 owing to
noise problems on 20 mx from ptc which is going to require a complete
station rework-earthing etc etc probably plus ferrite beading and other
shutting up measures. Since I am going to be moving qth before to much
longer this can wait till then. I have been active on pacsats and 9600 bd
sats this is a new and welcome (and interesting) change in direction.
If anybody has any mods to improve paccom ptc 1's I would be grateful for
any info.
I have interent WWW access and have accessed Phil Suusmans page (and
subscribed) as well as SCS .
Hope to meet some of you via this or on HF 'ere long-be hapy and well
Graham at Dromedary near Hobart Tasmania , Australia.
email vk7zo@ calvados.apana.org.au
packet vk7zo@vk7gl.#hbt.tas.oc.au
73's

--
vk7zo@calvados.apana.org.au (Graham Ranft)
APANA, Hobart, Tasmania: 03 6234 8530

120. Re: [HFSIG:1457] Re: EVM56002 ?

From: Greg Jones <wd5ivd@tapr.org>
Date: Sun, 25 Aug 1996 10:28:02 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608251528.KAA02689@tapr.org>
Rendered body type: text/plain

Yes -- this is a great offer. Dorothy has about 40 on order right now.

Please pass the word to students and school, since they can get in on the
offer as well.

I am hoping this is one of many future involvements with Motorola in the
future.

Cheers - Greg, WD5IVD

>
> Greg Jones wrote:
>
> > There is an opportunity for TAPR to do a group purchase on the Motorola
> > EVM56002 at a pretty low cost.   Between 20 and 200 units would be the
> > goal...same price no matter how many are purchased. Figure about $60 under
> > current commercial prices. No hard prices, becuase more work would have to
> > be done.
> >
> > Would there be interest in this in the HF SIG ? Espiecally since Johan's
> > code would run on it.
>
> Sounds great considering the price is already only $149.95 at the
> Design-Net price page (http://www2.motorola-dsp.com/dsp/home/net/pg.html).
>
> They also offer a student discount of 20% reducing the price to $120 + $3
> shipping (http://www2.motorola-dsp.com/dsp/home/edu/discount.html)
>
> Mike N4CNW
>
>

121. Re: [HFSIG:1517] Re: EVM56002 ?

From: forrerj@peak.org (Johan Forrer.)
Date: Sun, 25 Aug 1996 21:30:58 -0800
To: hfsig@tapr.org
Message-ID: <199608260439.VAA01385@PEAK.ORG>
Rendered body type: text/plain

Hi Greg,

Sounds like things are going with the EVM orders - I'm pleased to hear that.
I don't think anyone will be dissapointed as it is a great DSP to work with;
a lot of PD code and development tools available too. I suspect that by time
of the DCC you may even have further orders.

Looking forward to the DCC. My wife will be coming along this time, and we
will be in Seattle already by Thursday to visit a bit. One of our favorite
cities.

I need to mention one item for the HFSIG agenda that we have to take care of
at the DCC; that is that we need to find someone that would be willing to
fill the function of moderator/scribe/organiser. Perhaps it is a good idea
to get some fresh ideas. I will continue my participation and sharing in my
experimental and development efforts.

--Johan

>Yes -- this is a great offer. Dorothy has about 40 on order right now.
>

>Please pass the word to students and school, since they can get in on the
>offer as well.
>
>I am hoping this is one of many future involvements with Motorola in the
>future.
>
>Cheers - Greg, WD5IVD

122. Who sells HF path simulators and related equipment?

From: Rob Roy <rroyael@laser.net>
Date: Tue, 27 Aug 1996 13:26:35 -0500
To: hfsig@tapr.org
Message-ID: <2.2.16.19960827182635.315fb6f4@smtp1.laser.net>
Rendered body type: text/plain

Hi;

  I realize that there are a lot of different HF path simulators
that are used for testing. I would like to obtain any and all
information on current/future systems. The results of this
query will be posted.

--
                    Rob Roy
rroyael@laser.net

123. Re: [HFSIG:1519] Who sells HF path simulators and related equipment?

From: forrerj@peak.org (Johan Forrer.)
Date: Tue, 27 Aug 1996 13:52:09 -0800
To: hfsig@tapr.org
Message-ID: <199608272100.OAA01809@PEAK.ORG>
Rendered body type: text/plain

Rob,

I cannot speak for professional equipment manufacturers - there are several
and they are expensive ~ $30,000 range for a Watterson channel simulator.

However, a blatant plug for the work and interests from this list (HFSIG) -
we are proud of it.

HFSIG have been discussing HF channel simulators based on the Watterson
ionospheric model for some time (approximately 3 years). This model
represents narrow-band ionospheric propagation behavior as it affects HF,
typically an SSB bandwidth channel (ca. 3000 Hz) - typically to conform to
test conditions as set out by the CCIR. This includes various types/degrees
of channel disturbances, such as multipath, flat, and selective fading.
Although of much importance, this does not include simulated noise - that is
the subject of further work.

The objective and incentive for this work was to provide a low-cost
implementation for amateurs to evaluate modems for HF use. Based on work
initiated by Alexander Kurpiers, DL8AAU, and with generous assistance from
Juergen Hash, DG1SCR, I made a simplified version of such a simulator. This
particular implementation runs on TAPR's DSP-93, which uses a TMS320C25 DSP
chip. Executable code is available for downloading (free for amateur use) on
the HFSIG files area. Contact TAPR for further details on the DSP-93
hardware. It may be a good idea to get in touch with Alexander should you
want to know more about this particular implementation.

You may also want to browse through the work done by Eric Johnson for NTIA
regarding ALE (atanasoff.nmsu.edu/pub/hf). I have been told that Eric does
related consulting work and offers a low-cost HF channel simulator.

As for the future - I am considering developing a professional version HF
channel simulator to compete with the $30,000 Watterson channel simulator
hardware. E-mail me for further information if you are interested.

Hope this is of interest.

--Johan Forrer
(e-mail: forrerj@peak.org)

>Hi;
>
> I realize that there are a lot of different HF path simulators
>that are used for testing. I would like to obtain any and all
>information on current/future systems. The results of this
>query will be posted.
>
>--
>   Rob Roy
>rroyael@laser.net
>
>
>

124. HF Channel Simulator

From: ejohnson@NMSU.Edu ()
Date: Wed, 28 Aug 1996 09:01:10 -0600
To: hfsig@tapr.org
Message-ID: <199608281501.JAA18442@NMSU.Edu>
Rendered body type: text/plain

Hi,

Tim Baggett (a former student) forwarded your request for info about
HF channel simulators. By mid-September I'll be shipping a new
low-cost Watterson model simulator based on the Motorola 56002.

It simulates two channels simultaneously, each having two independently-fading paths. The user interface is a serial port, through which you can set (for each channel):

 * the SNRs for the two paths
 * the Doppler spread (fading bandwidth)
 * the delay between the two paths.

Since this was originally developed for manufacturers of MIL/FED-STD ALE gear, there is also a set of preset combinations for Gaussian noise, ITU-Good, and ITU-Poor channels at specific SNRs.

The selling price for just the DSP board (unencumbered by case, power supply, etc.) is $500. A unit mounted in a case with power supply is $1000.

Contact: Johnson Research
100 Horseshoe Circle
Las Cruces, NM 88005

Regards

Eric Johnson

September 1996

Original source file: hfsig.9609
Source: https://archive.org/details/tapr-1997
Recovered from: https://archive.org/download/TAPR-HF_1996-09/TAPR-HF_1996-09.txt

1. (no subject)

From: dona@tridelta.com (Don Adams)
Date: Thu, 5 Sep 1996 10:19:11 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199609051419.KAA22674@sparcy.tridelta.com>
Rendered body type: text/plain

I've been searching the internet for a few evenings.
 I found some abstracts
at http://sun4.iaee.tuwien.ac.at/jb9495/ber20002.html
 http://www.ee.ed.ac.uk/profiles/research/SaS/SaS-info.html
that were interesting concerning a RAKE tracking filter.(?!)
It claims to be a filter or corrolator
that needed "no prior knowledge of the signal". and can
correct multipath distortion.

Does anyone know what this really is?

Over and Out Don

2. Re: [HFSIG:1522]

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 5 Sep 1996 11:54:43 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199609051854.LAA12594@servo.qualcomm.com>
Rendered body type: text/plain

Don,

I haven't looked at the article yet, but a RAKE receiver is a multichannel
spread spectrum receiver that tracks the individual multipath components
of a signal and then combines them before demodulation. We rely pretty
heavily on them in Qualcomm's IS-95 CDMA system.

The basic idea of a RAKE dates from the late 1950s.

To resolve individual multipath components, the time delay between
them must be at least one chip. This corresponds to a spreading
bandwidth equal to or greater than the reciprocal of the delay.

Alternatively, this corresponds to the spreading bandwidth being
greater than the coherence bandwidth of the channel. Think of the
channel as imposing a time-varying comb filter on your signal. If the
signal is wide compared to the frequency interval between the "teeth"
of the comb, then the comb is very unlikely to remove all of the
signal at once. This is frequency diversity in action.

I don't know of any way that a RAKE receiver can be used on a
narrowband signal. If the narrowband signal falls into one of the
notches in the comb, nothing can bring it back. You then need other
forms of diversity (time, space).

Phil

3. Johan's modem

From: Robert.Glassey@nmp.nokia.com
Date: Mon, 9 Sep 96 18:09:52 +0100
To: hfsig@tapr.org
Message-ID: <H0000292026184l5@MHS>
Rendered body type: text/plain

Hi,

I downloaded the WAV files of Johan's modem waveforms. I've been able to
load them into matlab, and I can see the 4 modulated tones, and I can
see amplitude and phase modulation on each one, but I cannot extract any
symbol timing or see any obvious constalation, no mater how I vary the
reference carrier frequency. I can see paterns that repeat after a short
time, but can't make any sense out of them. It would help if the data
was longer and more random.

Can you demodulate this waveform? if so how? What do you lock to? Is
there a proper constalation?

Cheers,

Rob, G0VTQ

4. Re: [HFSIG:1524] Johan's modem

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 10 Sep 1996 09:20:30 -0800
To: hfsig@tapr.org
Message-ID: <199609101626.JAA20048@PEAK.ORG>
Rendered body type: text/plain

>Hi,
>
>I downloaded the WAV files of Johan's modem waveforms. I've been able to
>load them into matlab, and I can see the 4 modulated tones, and I can
>see amplitude and phase modulation on each one, but I cannot extract any
>symbol timing or see any obvious constalation, no mater how I vary the
>reference carrier frequency. I can see paterns that repeat after a short
>time, but can't make any sense out of them. It would help if the data
>was longer and more random.
>
>Can you demodulate this waveform? if so how? What do you lock to? Is
>there a proper constalation?
>
>Cheers,
>
>Rob, G0VTQ
>
>

Hi Rob,

Pleased to see your interest in the waveforms. A few suggestions that may
put you on the right track:

I do have demodulators that goes with each of the sample waveforms, so rest
assured, the signals can be demodulated and produce text-book constellations.
All the waveforms are QPSK, so you need a QPSK demodulator(s). You may use
coherent or non-coherent demodulation, and as the S/N is very good, you
should have no problem to phase lock - at least I that is my experience. For
detection, a matched filter works best, also keep in mind that these are
RC-shaped pulses.

All waveforms has
        a) a pre-amble, where the phase is stepped 90 degree per symbol (75
baud rate),
        b) followed by a number of cycles 000/001/010/011/100/....

Since I have provided the baseband waveforms, here are the carrier tones that I
used:

```
Waveform 1: 1350 Hz              ( single channel   DQPSK)
         2: 1350/1500 Hz         ( dual   channel   DQPSK)
         3: 1350/1500/1650/1800  ( quad   channel   DQPSK -- QUATOR)
```

As you may notice form above, the tones are spaced at 2*Baudrate and there
are eight phases on each carrier tone.

I would suggest you start off with the single channel waveform, extract and
study the I/Q waveform relationship, especially the pre-amble which will
reveal a lot.

For folks attending the DCC next week; a brief overview of QUATOR is planned
for the HFSIG meeting. I'll show some ideas of how the bit assembly is to be
done for achieving the time/frequency diversity. Keep in mind that things
are experimental and things are wide open for suggestions/critisism. This is
your opportunity to bring your ideas for further discussion.

As far as the actual implementation, at least what I am doing; I have been
using the PSA sound card platform thus far. This seems to have worked out
exteremely well for coding various parts of the modem, i.e., for monitoring
the workings of things like Costas loops and bit sync algorithms.
For the future, however, I'm not sure which way I'll go. My experience with
the new TI 320C31 DSK has been very encouraging - perhaps too little memory,
but it is really nice for the task, the speed and dynamic range, and a high
speed connection with the PC. This would make an ideal free-standing

implementation. On the other hand, I'm tempted to develop on the PC using
the soundblaster, but that would have no chance of ever getting the the ARQ
mode to work. I will, however, be able to play with the broadcast and CSMA
modes.
I'll need to see how things develop - too much going on at the moment and
the new modem have been on the shelf due to that. Would be nice to resume
experimental work again.

Much rambling, but hope this helps,

--Johan

5. coding progress

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 10 Sep 1996 16:03:20 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199609102303.QAA07499@servo.qualcomm.com>
Rendered body type: text/plain

Hello all,

I've been rather quiet here of late so I thought I'd mention what I've
been doing in my spare time over the past couple of weeks.

I've continued to fine-tune my Viterbi decoder for speed and Eb/N0
performance.

My previous version uses the "register exchange" method to update the
path histories. This limited the path history to 32 bits, the size of
a long integer in C. This is barely adequate as the rule-of-thumb says
that the path history should be at least 4-5 times the constraint length
(K=7), longer if the code is going to be punctured to high rates.

I rewrote my current version to use the traceback method. In this
case, the path histories are saved all the way from the beginning of
the packet. At the end of the packet, I "trace back" through the path
memory to determine the decoded data. Although this version is
packet-oriented at present, it shouldn't be too hard to modify to
operate in a continuous stream mode where traceback is done every N
bits (N >> K).

I've also spent quite a bit of effort tuning up the Viterbi decoder
for speed. Through a few more C coding tricks, I now have it running
at 258 kb/s on a 133 MHz Pentium. (The compiler is GCC 2.7.2 with full
optimization).

Thanks to my success with fast Viterbi decoding I've gotten more
interested of late in concatenated Reed-Solomon/Viterbi coding. These
codes were used on the Voyager mission starting at Uranus, and they
are now an integral part of digital satellite broadcasting (European
DVB, US DSS).

In terms of Eb/N0 they perform about as well as sequential decoding
but with more uniform computational effort. The BER curve for the
Voyager code is a nearly vertical wall at a Eb/N0 of about 2.5 dB.
The Voyager outer code is a (255,223) RS code over GF(256), and the
inner code is a rate 1/2 K=7 Viterbi-decoded convolutional code. DVB
uses a shortened (204,188) RS outer code and a K=7 Viterbi inner code
at various rates from 1/2 to 7/8, so it's not quite as strong as the
Voyager code but is still quite good.

This has gotten me interested in Reed-Solomon codes. I found some
freely available C code on the net by Robert Morelos-Zaragoza and Hari
Thirmoorthy of the University of Hawaii that does errors+erasures
decoding of Reed-Solomon codes. I've been massaging it for speed and
interface cleanliness. I now have it decoding the Voyager (255,223) RS
code over GF(256) at about 850 kb/s with a full load of errors and
erasures, and about twice that speed when the frame has no errors.

I'll put both my new Viterbi decoder and the RS stuff up on my web page
in a few days when I package it up.

Phil

6. Re: [HFSIG:1525] Re: Johan's modem

From: Jim Sanford <wb4gcs@amsat.org>
Date: Tue, 10 Sep 1996 20:33:50 -0400
To: hfsig@tapr.org

Message-ID: <1.5.4.32.19960911003350.8c5091dc@infi.net>
Rendered body type: text/plain

At 11:28 AM 9/10/96 -0500, you wrote:
>>Hi,
>>
>>I downloaded the WAV files of Johan's modem waveforms. I've been able to
>>load them into matlab, and I can see the 4 modulated tones, and I can
>>see amplitude and phase modulation on each one, but I cannot extract any
>>symbol timing or see any obvious constalation, no mater how I vary the
>>reference carrier frequency. I can see paterns that repeat after a short
>>time, but can't make any sense out of them. It would help if the data
>>was longer and more random.
>>
>>Can you demodulate this waveform? if so how? What do you lock to? Is
>>there a proper constalation?
>>
>>Cheers,
>>
>>Rob, G0VTQ
>>
>>
>
>Hi Rob,
>
>Pleased to see your interest in the waveforms. A few suggestions that may
>put you on the right track:
>
>I do have demodulators that goes with each of the sample waveforms, so rest
>assured, the signals can be demodulated and produce text-book constellations.
>All the waveforms are QPSK, so you need a QPSK demodulator(s). You may use
>coherent or non-coherent demodulation, and as the S/N is very good, you
>should have no problem to phase lock - at least I that is my experience. For
>detection, a matched filter works best, also keep in mind that these are
>RC-shaped pulses.
>
>All waveforms has
>   a) a pre-amble, where the phase is stepped 90 degree per symbol (75
>baud rate),
>   b) followed by a number of cycles 000/001/010/011/100/....
>
>Since I have provided the baseband waveforms, here are the carrier tones that I
>used:
>
>Waveform 1: 1350 Hz   (single channel   DQPSK)
>   2: 1350/1500 Hz   (dual   channel   DQPSK)
>   3: 1350/1500/1650/1800 (quad   channel   DQPSK -- QUATOR)
>
>As you may notice form above, the tones are spaced at 2*Baudrate and there
>are eight phases on each carrier tone.
>
>I would suggest you start off with the single channel waveform, extract and
>study the I/Q waveform relationship, especially the pre-amble which will
>reveal a lot.
>
>For folks attending the DCC next week; a brief overview of QUATOR is planned
>for the HFSIG meeting. I'll show some ideas of how the bit assembly is to be
>done for achieving the time/frequency diversity. Keep in mind that things
>are experimental and things are wide open for suggestions/critisism. This is
>your opportunity to bring your ideas for further discussion.
>
>As far as the actual implementation, at least what I am doing; I have been
>using the PSA sound card platform thus far. This seems to have worked out
>exteremely well for coding various parts of the modem, i.e., for monitoring
>the workings of things like Costas loops and bit sync algorithms.
>For the future, however, I'm not sure which way I'll go. My experience with
>the new TI 320C31 DSK has been very encouraging - perhaps too little memory,
>but it is really nice for the task, the speed and dynamic range, and a high
>speed connection with the PC. This would make an ideal free-standing
>implementation. On the other hand, I'm tempted to develop on the PC using
>the soundblaster, but that would have no chance of ever getting the the ARQ
>mode to work. I will, however, be able to play with the broadcast and CSMA
>modes.
>I'll need to see how things develop - too much going on at the moment and
>the new modem have been on the shelf due to that. Would be nice to resume
>experimental work again.
>
>Much rambling, but hope this helps,
>
>--Johan

>
>
Johan:
PLEASE stay with the PSA chip set . . . .
73, Jim
wb4gcs@amsat.org
>

7. Re: [HFSIG:1526] coding progress

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 10 Sep 1996 19:26:30 -0800
To: hfsig@tapr.org
Message-ID: <199609110232.TAA27279@PEAK.ORG>
Rendered body type: text/plain

Phil,

Nice progress - I am looking forward to downloading the latest versions of
the decoders.

A bit of feedback if you're interested - I have had a bash at converting the
Fano decoder to run under MS VC++ version 4.0. This is for the NT/WIN95
32-bit platform. I found that there are a number of routines that are common
UNIX or GCC, but are missing from the MS libraries. I'll post something
about this, but perhaps see your latest version first.

Keep up the good work!

--Johan

>Hello all,
>
>I've been rather quiet here of late so I thought I'd mention what I've
>been doing in my spare time over the past couple of weeks.
>
>I've continued to fine-tune my Viterbi decoder for speed and Eb/N0
>performance.
>
>My previous version uses the "register exchange" method to update the
>path histories. This limited the path history to 32 bits, the size of
>a long integer in C. This is barely adequate as the rule-of-thumb says
>that the path history should be at least 4-5 times the constraint length
>(K=7), longer if the code is going to be punctured to high rates.
>
>I rewrote my current version to use the traceback method. In this
>case, the path histories are saved all the way from the beginning of
>the packet. At the end of the packet, I "trace back" through the path
>memory to determine the decoded data. Although this version is
>packet-oriented at present, it shouldn't be too hard to modify to
>operate in a continuous stream mode where traceback is done every N
>bits (N >> K).
>
>I've also spent quite a bit of effort tuning up the Viterbi decoder
>for speed. Through a few more C coding tricks, I now have it running
>at 258 kb/s on a 133 MHz Pentium. (The compiler is GCC 2.7.2 with full
>optimization).
>
>Thanks to my success with fast Viterbi decoding I've gotten more
>interested of late in concatenated Reed-Solomon/Viterbi coding. These
>codes were used on the Voyager mission starting at Uranus, and they
>are now an integral part of digital satellite broadcasting (European
>DVB, US DSS).
>
>In terms of Eb/N0 they perform about as well as sequential decoding
>but with more uniform computational effort. The BER curve for the
>Voyager code is a nearly vertical wall at a Eb/N0 of about 2.5 dB.
>The Voyager outer code is a (255,223) RS code over GF(256), and the
>inner code is a rate 1/2 K=7 Viterbi-decoded convolutional code. DVB
>uses a shortened (204,188) RS outer code and a K=7 Viterbi inner code
>at various rates from 1/2 to 7/8, so it's not quite as strong as the
>Voyager code but is still quite good.
>
>This has gotten me interested in Reed-Solomon codes. I found some
>freely available C code on the net by Robert Morelos-Zaragoza and Hari
>Thirmoorthy of the University of Hawaii that does errors+erasures
>decoding of Reed-Solomon codes. I've been massaging it for speed and
>interface cleanliness. I now have it decoding the Voyager (255,223) RS
>code over GF(256) at about 850 kb/s with a full load of errors and

>erasures, and about twice that speed when the frame has no errors.
>
>I'll put both my new Viterbi decoder and the RS stuff up on my web page
>in a few days when I package it up.
>
>Phil
>
>

8. Re: coding progress

From: Eduardo Jacob <edu@kender.es>
Date: Wed, 11 Sep 1996 20:38:41 -0100
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19960911213841.0067b2b4@kender.es>
Rendered body type: text/plain

>A bit of feedback if you're interested - I have had a bash at converting the
>Fano decoder to run under MS VC++ version 4.0. This is for the NT/WIN95
>32-bit platform. I found that there are a number of routines that are common
>UNIX or GCC, but are missing from the MS libraries. I'll post something
>about this, but perhaps see your latest version first.

I have read in DDJ that there is a port of GNU-C to Windows'95,
perhaps you could use it. I don't remember very well but I think the missing
parts were those related to the GUI. For dos perhaps DDJ could suit you.

Eduardo EA2BAJ
Eduardo, EA2BAJ

9. Re: [HFSIG:1529] Re: coding progress

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 11 Sep 1996 15:40:58 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199609112240.PAA18109@servo.qualcomm.com>
Rendered body type: text/plain

>>A bit of feedback if you're interested - I have had a bash at converting the
>>Fano decoder to run under MS VC++ version 4.0. This is for the NT/WIN95
>>32-bit platform. I found that there are a number of routines that are common
>>UNIX or GCC, but are missing from the MS libraries. I'll post something
>>about this, but perhaps see your latest version first.

I'm a little confused by this, as the Fano decoder is just a low level
subroutine that doesn't make use of anything elaborate in the
operating environment. It should port to just about any C environment.

> I have read in DDJ that there is a port of GNU-C to Windows'95,
>perhaps you could use it. I don't remember very well but I think the missing
>parts were those related to the GUI. For dos perhaps DDJ could suit you.

There is a port of GNU-C to DOS called DJGPP that I'm pretty familiar with.
A few weeks ago I released a port of KA9Q NOS to DJGPP; it's up on my web
page. Are you saying there is specific support for windows 95 in graphics
(not dos) mode?

Phil

10. Re: [HFSIG:1530] Re: coding progress

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 11 Sep 1996 23:30:28 -0800
To: hfsig@tapr.org
Message-ID: <199609120636.XAA12565@PEAK.ORG>
Rendered body type: text/plain

Hi Phil,

Here are more specifics that may help;

>>>A bit of feedback if you're interested - I have had a bash at converting the
>>>Fano decoder to run under MS VC++ version 4.0. This is for the NT/WIN95
>>>32-bit platform. I found that there are a number of routines that are common
>>>UNIX or GCC, but are missing from the MS libraries. I'll post something
>>>about this, but perhaps see your latest version first.
>
>I'm a little confused by this, as the Fano decoder is just a low level
>subroutine that doesn't make use of anything elaborate in the
>operating environment. It should port to just about any C environment.
>

I'm using VC version 4.0 (4.2 is the latest):

Metrics.c: erf() .... not recognized perhaps there is an equivalent.

M_SQRT2, M_LOG2E .... not recognized - these are UNIX/GCC natives.

Sim.c: random() ... does'nt recognize this either.

Of course, curses does'nt exsist, but one can work around that. Otherwise
the function you use to parse your command-line arguments also doesn't have
an equivalent. There are a few minor picky warnings regarding type casting
that I agree with the compiler, but won't hurt.

>> I have read in DDJ that there is a port of GNU-C to Windows'95,
>>perhaps you could use it. I don't remember very well but I think the missing
>>parts were those related to the GUI. For dos perhaps DDJ could suit you.
>
>There is a port of GNU-C to DOS called DJGPP that I'm pretty familiar with.

Yes - the makefile for the Fano decoder and test programs all compile/link
and execute 100% with DJGPP - either under DOS or a DOS box under WIN95.

>A few weeks ago I released a port of KA9Q NOS to DJGPP; it's up on my web
>page. Are you saying there is specific support for windows 95 in graphics
>(not dos) mode?
>
>Phil
>
>

I suspect he might be referring to that it is for generating Win95 "console"
mode code, i.e., 32-bit programs that run in a Win95 DOS box. These do not
use the Windows GUI functions.

DJGPP is very slick and I like it a lot. My main objection with DJGPP,
however, is when one venture into doing hardware access, which is what we
used to used to do with DOS programs - DJGPP does that with with nasty old
DPMI calls that I really discourage anyone to use. I personally feel that I
would want to move away from DOS and things like DOS extenders and the
limitations of 64K segments. The right way, in my humble opinion, is to bite
the bullet and write your own VxDs to do that kind of thing using the native
mechanisms provided by the operating system. This way one can control and
guarantee interrupt latency and control the flow of data/control between
Win95-based programs and real-time, low-level, device dependant code. Not
that I think Win95 is all that great a piece of code, but I suspect that any
code that will run under that platform will enjoy that much greater appeal
in the ham community and get folks interested. If that was'nt the case, I
would prefer to do all my work under Linux
(my two-cents worth).

--Johan

11. Re: [HFSIG:1530] Re: coding progress

From: jmorriso@bogomips.com (John Paul Morrison)
Date: Thu, 12 Sep 1996 00:18:28 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <m0v162e-000TyBC@bogomips.com>
Rendered body type: text/plain

>
> I'm a little confused by this, as the Fano decoder is just a low level
> subroutine that doesn't make use of anything elaborate in the
> operating environment. It should port to just about any C environment.
>
> > I have read in DDJ that there is a port of GNU-C to Windows'95,
> >perhaps you could use it. I don't remember very well but I think the missing
> >parts were those related to the GUI. For dos perhaps DDJ could suit you.
>
> There is a port of GNU-C to DOS called DJGPP that I'm pretty familiar with.
> A few weeks ago I released a port of KA9Q NOS to DJGPP; it's up on my web
> page. Are you saying there is specific support for windows 95 in graphics
> (not dos) mode?

Cygnus support sponsored a port to Windows NT, plus compatibility
libraries to make UNIX programs easy to port. Windows 95 implements
some of the Windows NT (Win32) API, so it runs under that too

(but not as well apparently).

I think it can only do console mode (ie text), more code needs
to be written so GCC can create GUI apps.

>
> Phil
>
>

BogoMIPS Research Labs -- bogosity research & simulation -- VE7JPM --
  jmorriso@bogomips.com ve7jpm@ve7jpm.ampr.org jmorriso@ve7ubc.ampr.org
---

12. Re: coding progress

From: "Eduardo Jacob" <jtpjatae@bicc00.bi.ehu.es>
Date: Thu, 12 Sep 1996 12:47:06 +0000
To: hfsig@tapr.org
Message-ID: <9609121149.AA21376@bicc00.bi.ehu.es>
Rendered body type: text/plain

> >   I have read in DDJ that there is a port of GNU-C to Windows'95,
> >perhaps you could use it. I don't remember very well but I think the missing
> >parts were those related to the GUI. For dos perhaps DDJ could suit you.
>
> There is a port of GNU-C to DOS called DJGPP that I'm pretty familiar with.
> A few weeks ago I released a port of KA9Q NOS to DJGPP; it's up on my web
> page. Are you saying there is specific support for windows 95 in graphics
> (not dos) mode?

   For DOS I was speaking about DJGPP, not DDJ. Yes, the objective is
to be able to build full GUI programs on Windows95 and NT. But I
think it is not totally working. They are on Beta 16 There is
information in

        ftp://ftp.cygnus.com/pub/sac/gnuwin32

   I have read it in DrDobbs Journal pages 121-126.

    Eduardo EA2BAJ
Eduardo Jacob - Area de Ingenieria Telematica
Departamento de Electronica y Telecomunicaciones
ETSII y de IT   Tel: +34-(9)4-427 8055
UPV / EHU   Fax: +34-(9)4-441 4041
Alda Urquijo s/n   E-mail: jtpjatae@bi.ehu.es
E-48013 - Bilbao (Spain)   : 100021,2212 Compuserve

13. Re: [HFSIG:1526] coding progress

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Thu, 12 Sep 1996 07:55:50 -0400
To: hfsig@tapr.org
Message-ID: <3237FA46.2C96@worldnet.att.net>
Rendered body type: text/plain

Phil Karn wrote:
>
> Hello all,
>
> I've been rather quiet here of late so I thought I'd mention what I've
> been doing in my spare time over the past couple of weeks.
>
> I've continued to fine-tune my Viterbi decoder for speed and Eb/N0
> performance.
>

I saw a brilliant talk by Bob McEliece yesterday at the retirement bash
for
Neal Zierler. It was on Turbo Codes. I have a good understanding of
what
they are about now, maybe even better than McEliece ;-). He has built
an
intuitive understanding of WHY they work but cannot prove it. It is
closely
enough related to some stuff that I have done that I believe I CAN prove
why they work and I have some techniques at my disposal that will allow
me
to accelerate their convergence. He claims that Viterbi claims they are

the most significant development in N years (where N closely matches the
number of years since he development the Viterbi decoder). You and I
have
had a conversation before on the telephone where I told the win of
a posteriori estimation over dynamic programming based estimation. Let
me
reemphasize it here since it is at the very heart of Turbo codes.
Dynamic
programming (Viterbi decoding) finds the best PATH through the possible
paths of symbols. What you want is to minimize the probability of
making
an error at each decision time given all the available evidence. It is
this that Hidden Markov Models do (where the symbol is the Markov chain
and the noisy channel is providing the Hidden ;-). Turbo codes have at
their heart, HMM, and the decoder approximates the EM algorithm (which
was first developed here at my company and the first rigorous proofs
concerning it were made by Baum, Petrie, Soules, and Welch while working
here). My experiments show that Turbo codes get <<<<<< within 1 dB of
the Shannon limit and indeed get in a case I have tried to -0.6 dB Eb/N0
>>>>>>>. You must climb onto the power curve on this it IS the future
of coding. I posted this to everyone since you appear to be ham radio's
primary instructor on coding these days and everyone could benefit by
discussions.

Bob
****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio. Scouts: Council Commissioner
*
* East Windsor, NJ 08520   * Post 995 Advisor, Troop 5700 Advancement,
*
* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061   * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240   * Council. Brownie Troop 193
*
****

14. Re: [HFSIG:1531] Re: coding progress

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 12 Sep 1996 11:22:38 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199609121822.LAA28123@servo.qualcomm.com>
Rendered body type: text/plain

>Metrics.c: erf() .... not recognized perhaps there is an equivalent.
>
>   M_SQRT2, M_LOG2E .... not recognized - these are UNIX/GCC natives.

>Sim.c: random() ... does'nt recognize this either.

Ah, those are both support files, not the actual decoder...

erf() is in the UNIX math library, and should be in most PC subroutine
libraries. It's the gaussian error function and is used to build the
metric tables from "first principles", i.e., from a knowledge of the
channel transition probabilities that are a function of Eb/N0 and amplitude.

M_SQRT2 and M_LOG2E are constants usually defined in <math.h>. M_SQRT2
is sqrt(2) and M_LOG2E is log2(e).

random() is a random number generator returning a value between 0 and
0x7fffffff inclusive. It's used to generate random bits and gaussian
noise for test purposes.

I note that the metric table can be built "offline" e.g., on a UNIX
system or under DJGPP with a math package and then the table(s)
included as compile-time data in the final program. Although the Fano
decoder works best when you compute a metric table for the actual
Eb/N0, I've found experimentally that you can use a fixed table
computed for the worst Eb/N0 you can handle (the computational cutoff
rate, about 2.5 dB for rate 1/2) and then scale the incoming signal to
maintain the constant noise level used to build the metric table. Much
stronger signals will saturate the soft-decision samples and
effectively degrade the decoder to hard decision decoding, but if
they're that strong and clean in the first place that won't hurt
anything. And signals noiser than 2.5 dB, of course, won't decode

anyway.

In other words, don't worry about the signal amplitude, it's the noise amplitude that counts.

That's because the most important thing with the fano decoder is that the branch metrics on the correct path be generally positive. If the signal level is too low for the metric table in use, then the decoder will constantly back up thinking it has made an error when in fact it's on the best path. That's why it's important to scale the incoming signal to attempt to keep the noise level constant. That also explains why a signal AGC is a *very bad* idea in Fano decoding, if it causes the noise level to vary. You want to keep the noise level constant and let the signal level vary if it has to. Again, let the signal saturate if it gets strong, that won't hurt anything.

One of the big advantages of the Viterbi decoder over the Fano decoder, btw, is that the Viterbi decoder works for just about any metric table, as long as it's not "upside down", i.e., as long as its matched to the signal polarity. Even the set of integers 0-255 seems to work well. As long as the signal amplitude is such as to use at least a few bits of your sample dynamic range, the algorithm will function reasonably optimally.

Phil

15. Re: [HFSIG:1532] Re: coding progress

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 12 Sep 1996 11:24:51 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199609121824.LAA28126@servo.qualcomm.com>
Rendered body type: text/plain

Thanks for the info on GCC under windows. Of course, the question remains: is it possible to write anything approaching "real time" code under that system, as opposed to running under DOS on a bare machine? You can alleviate the problem with buffering (e.g., on a sound card). But I've already seen at least one machine running W95 (my laptop) that stutters occasionally like Max Headroom when playing back audio, probably because of missed interrupts. It never did that under W 3.1.

Phil

16. Re: [HFSIG:1534] Re: coding progress

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 12 Sep 1996 11:29:23 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199609121829.LAA28130@servo.qualcomm.com>
Rendered body type: text/plain

I agree, turbo codes are very exciting and they probably represent coding's last gasp. I've heard both Viterbi and McEliece speak at Qualcomm on this subject.

But there's a catch. Patents are flying in this area faster than bullets in a battlefield, and since I'm interested in writing free software the last thing I want is to be sued. Fano decoders, Viterbi decoders, Reed-Solomon decoders and concatenated codes have been around for 25-35 years and are clearly in the public domain.

Phil

17. Re: [HFSIG:1532] Re: coding progress

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 12 Sep 1996 14:10:55 -0800
To: hfsig@tapr.org
Message-ID: <199609122117.OAA18381@PEAK.ORG>
Rendered body type: text/plain

John,

I checked this out and looks like an interesting development - they are at Beta 16, but still is a cross-port running on a Linux platform. The native compiler for the Win95 platform is still in it's infancy - will be a while if I interpret things correctly.

>> There is a port of GNU-C to DOS called DJGPP that I'm pretty familiar with.
>> A few weeks ago I released a port of KA9Q NOS to DJGPP; it's up on my web
>> page. Are you saying there is specific support for windows 95 in graphics
>> (not dos) mode?
>
>Cygnus support sponsored a port to Windows NT, plus compatibility
>libraries to make UNIX programs easy to port. Windows 95 implements
>some of the Windows NT (Win32) API, so it runs under that too
>(but not as well apparently).
>
>I think it can only do console mode (ie text), more code needs
>to be written so GCC can create GUI apps.
>
>>
>> Phil
>>
>>
>
>--
>BogoMIPS Research Labs -- bogosity research & simulation -- VE7JPM --
> jmorriso@bogomips.com ve7jpm@ve7jpm.ampr.org jmorriso@ve7ubc.ampr.org
>--
>
>

--Johan

18. Re: [HFSIG:1538] Re: coding progress

From: jmorriso@bogomips.com (John Paul Morrison)
Date: Thu, 12 Sep 1996 21:27:29 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <m0v1Pql-000TswC@bogomips.com>
Rendered body type: text/plain

>
> John,
>
>
> I checked this out and looks like an interesting development - they are at
> Beta 16, but still is a cross-port running on a Linux platform. The native
> compiler for the Win95 platform is still in it's infancy - will be a while
> if I interpret things correctly.
>
> --Johan

GCC is not yet self-hosting on Win95/WinNT (ie can recompile itself)
but it does compile programs (actually, I think it will build itself
with some manual assistance). Also, I don't think Windows 95 is the
primary goal of the Win32 port. It's meant to work with Windows NT;
the Windows 95 support is a secondary bonus because it supports
some of the Win32 api, but not others.

Depending on what you want to do, DJGPP programs work in Windows 95.
I was able to run the DJGPP version of NOS (which I cross-compiled in
Linux) in Windows 95, and use SLIP. Packet drivers should work too.

>
>

BogoMIPS Research Labs -- bogosity research & simulation -- VE7JPM --
  jmorriso@bogomips.com ve7jpm@ve7jpm.ampr.org jmorriso@ve7ubc.ampr.org
----

19. Re: [HFSIG:1536] Re: coding progress

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Fri, 13 Sep 1996 16:34:29 +0200
To: hfsig@tapr.org
Message-ID: <323970F5.35E5@ife.ee.ethz.ch>
Rendered body type: text/plain

> card). But I've already seen at least one machine running W95 (my
> laptop) that stutters occasionally like Max Headroom when playing back
> audio, probably because of missed interrupts. It never did that under
> W 3.1.
This seems to be a different problem. We're currently working
on Windows support for PC/FlexNet, and Baycom type modems and (U)SCC

cards at higher speeds run already, which have rather hard
requirements wrt interrupt latency.
Not that I am a Windoze advocat...

Tom

20. DCC 1996

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 23 Sep 1996 08:45:26 -0800
To: HFSIG@tapr.org
Message-ID: <199609231549.IAA27272@PEAK.ORG>
Rendered body type: text/plain

Hi All,

I will shortly post a short summary of my impressions of the recent DCC.

First; let me thank the hosts; BEARS (Boeing Employees Amateur Radio
Society) for all the kind and generous hospitality. The conference venue was
excellent, food great, and we were kept entertained and happy.

I also wish to thank ARRL and the folks from TAPR for all the hard work that
have gone into making such an event successful. Just to illustrate; the
proceedings consists of some 256 double-sided pages with excellent technical
content, for example, two outstanding contributions on 1.2 Mbit/s digital
tranceivers by Matjaz Vidmar, S53MV - complete with schematics.

The interest in HF digital was heartwarming and so was the HFSIG meeting (at
least from where I was standing on the other side of the podium!) The topic
this time was woven around past discussions we have had on HFSIG regarding
HF Channel simulation: Tom McDermot did an excellent presentation of the
mechnics involved in simulation using the Watterson model, that I followed
with showing a number of "doppler grams", courtesy of Peter Martinez, G3PLX.
These are really unique and worth seeing. A brief overview of software for
the DSP-93 and a demo running this implementation of the Watterson
ionospheric model, concluded this part of the meeting. I also presented an
outline of the work that I have been doing on QUATOR.

More to follow later on the DCC.

I also need to prepare a posting on some of the recent developments on the
slow-speed BPSK scene - this involves some innovative coding that are
especially suited for such narrow-band systems that are limited by human
typing speeds - quite contrary to the trend to building error-correcting
codes.

73's till later.

--Johan Forrer, KC7WW

21. G3PLX's Dopplergrams

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 24 Sep 1996 08:45:42 -0800
To: HFSIG@tapr.org
Message-ID: <199609241550.IAA09227@PEAK.ORG>
Rendered body type: text/plain

Hi,

I have had someone ask to post further details about how the dopplergrams
that I showed at the DCC were produced.

Actually it's an old trick used in monitoring meteor scatter and other weak
signals; monitor a known transmission that you know arrives via the ionosphere.
Peter's examples where recorded on 21 MHz, 17.7 MHz, 10.2 MHz, 7.7
MHz, and 3.5 MHz. Then a slice of the receiver's audio passband, say around
1 kHz are converted into a stream of I/Q values at 10 mS intervals and that
run through an FFT to produce the images. Peter then describes the display
as follow:

> "These images are time/frequency plots, with signal-level encoded
> as the brightness. They were originally 16-level, and the signal
> level is logarithmic with 3.75dB per grey-step, so that the full
> dynamic range is 60dB. In the vertical direction there are 256 pixels
> representing the frequency axis, and in most of the images there
> are 640 pixels horizontally for the time-axis."

The result appears like a strip chart with Doppler shift on the vertical
axis, and time along the horizontal axis. The doppler axis typically spans

256 pixels that represent either 25 Hz, 12.5 Hz, or 6.25 Hz Doppler, i.e., tracking some rather narrow frequency disturbances. The time axis typically spans over several hours, often a full day and night. In such cases band "closing" and "openings" are distinct and reveal several complex ionospheric mechanics such as meteor pings, sporadic E, F layer propagation, and rays splitting into (O)rdinary and (X)traordinary components. Even reflections from aircraft can be identified.

Peter indicated that he will be publishing some of this work and that would be something to look forward to. We are most grateful for his permission to show some examples of this work at the DCC.

--Johan

22. UUCP command execution failed

From: nuucp@ihig2.firewall.lucent.com
Date: Tue, 24 Sep 1996 11:58:36 -0500
To: hfsig@tapr.org
Message-ID: <199609241658.LAA05191@ihig2.firewall.lucent.com>
Rendered body type: text/plain

Your UUCP remote execution request 'igbAbb3e' (9/24-11:58:35)
failed on system 'ihig2'.
Your request: rmail bighorn.dr.lucent.com!grb
Reason for failure: command exited with exit code 1

23. Reed-Solomon encoder/decoder code available

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Fri, 27 Sep 1996 01:19:42 -0700 (PDT)
To: hfsig@tapr.org, tcp-group@ucsd.edu
Cc: robert@spectra.eng.hawaii.edu, harit@spectra.eng.hawaii.edu,
        simon@augean.eleceng.adelaide.edu.au
Message-ID: <199609270819.BAA16824@unix.ka9q.ampr.org>
Rendered body type: text/plain

Hi. I've just released the Reed-Solomon software I've been working on lately.

The software can encode and decode Reed-Solomon codes with symbol sizes ranging from 2^2 to 2^16 bits. The default parameters are for the (255,223) code with 8-bit symbols that was used on the Voyager-2 spacecraft at Uranus and Neptune.

The files are as follows:

http://www.qualcomm.com/people/pkarn/feccode/rs-readme
http://www.qualcomm.com/people/pkarn/feccode/rs.zip
http://www.qualcomm.com/people/pkarn/feccode/rs.tar.gz

I've also updated my ham radio digital communications page (http://www.qualcomm.com/people/pkarn/ham.html) with pointers to these files. Enjoy!

Phil

24. Re: [HFSIG:1544] Reed-Solomon encoder/decoder code available

From: Jim Sanford <jsanford@mailhost.infi.net>
Date: Fri, 27 Sep 1996 07:49:22 -0400
To: hfsig@tapr.org
Message-ID: <324BBF42.6A57@mailhost.norfolk.infi.net>
Rendered body type: text/plain

Phil Karn wrote:
>
> Hi. I've just released the Reed-Solomon software I've been working on
> lately.
>
> The software can encode and decode Reed-Solomon codes with symbol
> sizes ranging from 2^2 to 2^16 bits. The default parameters are for
> the (255,223) code with 8-bit symbols that was used on the Voyager-2
> spacecraft at Uranus and Neptune.
>
> The files are as follows:
>
> http://www.qualcomm.com/people/pkarn/feccode/rs-readme
> http://www.qualcomm.com/people/pkarn/feccode/rs.zip
> http://www.qualcomm.com/people/pkarn/feccode/rs.tar.gz
>
> I've also updated my ham radio digital communications page

> (http://www.qualcomm.com/people/pkarn/ham.html) with pointers to these
> files. Enjoy!
>
> Phil
Phil:
THANK YOU!

All the best 73, Jim wb4gcs@amsat.org

25. Re: [HFSIG:1544] Reed-Solomon encoder/decoder code available

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 27 Sep 1996 08:08:29 -0800
To: hfsig@tapr.org
Message-ID: <199609271512.IAA04514@PEAK.ORG>
Rendered body type: text/plain

Phil,

>Hi. I've just released the Reed-Solomon software I've been working on
>lately.

Great! I have just the application for it.

Thanks much.

--Johan

26. Re: [HFSIG:1544] Reed-Solomon encoder/decoder code available

From: Jim Sanford <jsanford@mailhost.infi.net>
Date: Fri, 27 Sep 1996 13:06:08 -0400
To: hfsig@tapr.org
Message-ID: <324C0980.14F1@mailhost.norfolk.infi.net>
Rendered body type: text/plain

>
> The files are as follows:
>
> http://www.qualcomm.com/people/pkarn/feccode/rs-readme
> http://www.qualcomm.com/people/pkarn/feccode/rs.zip
> http://www.qualcomm.com/people/pkarn/feccode/rs.tar.gz
>
> Phil
Phil:
What's the rs.zip and rs.tar.gz files -- different coding of same thing?
Netscape choked on the .gz file . . . .
  (Windoze version)

thanks, Jim

27. New Book

From: CDrentea@aol.com (by way of Rick Whiting <rwhiting@winternet.com>)
Date: Sun, 29 Sep 1996 20:04:42 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199609300104.UAA02558@icicle.winternet.com>
Rendered body type: text/plain

I came across a new book you may want to take a look at: Digital
Communications Techniques - Signal Design and Detection by Simon, Hinedi and
Lindsey - ISBN # 0-13-200610-3. It has been recently published by Prentice
Hall and costs about $82 - 10% from Barnes and Noble.

Very comprehensive. Although this information exists partially in other
books, the book presents the methods of coherent, noncoherent, partially
coherent, and differentially coherent detection for various binary and M-ary
coded and uncoded modulation techniques (BPSK, QPSK, MSK, MFSK, QASK, QAM,
etc) that can be applied to the design of uncoded systems while the
Ro-criterion (channel throughput in bits/sec/Hz) and bit error probability
are used as the methodology for designing efficient coded digital radio
communications circuits. The book has absolutely nothing on propagation
phenomena per say (it would be too good to be true), but presents the
communication channel methodically and much like an OSI-ISO seven layer
digital communications model (old stuff for data communications in WANs and
LANs, but a very "novel" idea for radio data communications).

Key concepts include scalar, vector, and waveform communications over the
additive nonwhite (colored) Gaussian noise channels. Communications via
Rayleigh and Rician fading (due to multipath) channels are also presented.

Enjoy!

28. Re: [HFSIG:1547] Re: Reed-Solomon encoder/decoder code available

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Mon, 30 Sep 1996 01:54:48 -0700 (PDT)
To: Jim Sanford <jsanford@mailhost.infi.net>
Cc: hfsig@tapr.org
Message-ID: <199609300854.BAA23664@unix.ka9q.ampr.org>
Rendered body type: text/plain

>What's the rs.zip and rs.tar.gz files -- different coding of same thing?

Yes. The rs.zip file is a PKZIP-format file (primarily for DOS users) and
the rs.tar.gz file is a GZIP-compressed tar archive (primarily for UNIX users).
I say "primarily" because tools to read each format do exist on both platforms.

Note that the test driver program is intended for a UNIX environment
and may require some tweaking to run under DOS, e.g, the library
routines called to generate random numbers may have to be changed.

The actual RS subroutines are self-contained and should compile
cleanly on any system that supports ANSI C.

Phil

October 1996

Original source file: hfsig.9610
Source: https://archive.org/details/tapr-1997
Recovered from: https://archive.org/download/TAPR-HF_1996-10/TAPR-HF_1996-10.txt

1. RELEASE OF BTL 1.0

From: Robert.Glassey@nmp.nokia.com
Date: Wed, 2 Oct 96 12:03:34 +0100
To: hfsig@tapr.org
Message-ID: <H00002920283aab6@MHS>
Rendered body type: text/plain

Blaster TeLetype V1.0
~~~~~~~~~~~~~~~~~~~~~~

Finally released and now available in the upload directory at URL:

ftp://ftp.tapr.org/tapr/SIG/hfsig/upload/btl10.zip

Blaster TeLetype (BTL) is my DSP RTTY demodulator package using the
Sound Blaster card (1.0 or greater) and a 386DX or greater IBM PC or
compatible.

BTL is FREE for amateur use. But volantary registration is encouraged to
support the ongoing developemnt of this program into a fully featured
DSP multimode modem using only a PC and sound card.

B T L 1.0 F E A T U R E S
~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~

o RX only RTTY demodualtion
o Fully adjustable centre frequency, with presets for standard tone
  sets.
o Fully adjustable frequency shift, 100Hz to 3000Hz, with easy presets.
o Fully adjustable baud rate, 25 to 200 Baud
o USB or LSB default Mark/Space polarity
o RXR Mark/Space tone reversal
o NAR Narrow/Wide tone filter option
o UNS Unshift on Space
o LOG Save received text to a log file
o 43/50 line mode
o Manually forced letters or figures shift
o Manually forced New Line on demod screen
o Demodulation pause key (Hold)
o On screen tuning and level indicators
o User adjustable Soundblaster input level settings and input selection
o User adjustable defaults and input levels saved in config file
o Support for Sound Blasters version 1.0 through AWE32.
o Sound Blaster parameter validation
o Extensive on-Line HELP!!
o Bright Colours!

Download and Enjoy!

Many thanks to the many HFSIG'er who have beta tested this software over
the past months. I an now fairly confident BTL is stable and works on
the vast majority of soundblasters, PC's and compatibles. No gaurantees
though, its free after all!

A description of the inards of BTL will follow soon.

Cheers,

Rob, G0VTQ

2. Development Environment

From: Lyle Johnson <lylej@azstarnet.com>
Date: Wed, 2 Oct 1996 07:34:23 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19961002073412.003524b4@azstarnet.com>
Rendered body type: text/plain

A couple years ago we bought the development system for the Star
Semiconductor SPROC chips. OF course, once we had the product designed and
debugged, Star went belly up and we had to switch horses :-)

The development system they sold consisted of a schematic interface (OrCAD).
The various building blocks in the schematic library were associatd with
pre-written and optimized code in both subroutine and in-line formats.
Arguments needed by the routines were entered onto the schematic as fields
(much as a resistor could have a "value" and a "tolerance", a signal
generaotr could have a "frequency" and an "amplitude."

Once you "sketched" your design, you'd tell the schematic capture tools to
output its "netlist", then turn the results of that loose on the "compiler"
which would look for syntax problems and, once it was happy, it would chrun
on the result and provide you with the otuptu code. You'd then download
this into the development lab and start debugging the design rather than the
details.

I guess loosely it is like using a high level language rather than assembler
to get things done.

Of course, the SPORC chip had superb real-time debugging characteristics,
but that is another story.

I'm wondering if anyone on this sig has the knowledge and inclination to do
something like this? Perhaps those that are good DSP coders could write the
various "modules" and someone strong in compiler knowledge could write the
tool that took the "netlist" and created the source module to run through
the mfr's assembler/linker/etc.

I could be persuaded to work on schematic symbols and so forth for at least
one or two popular (inexpensive?) windows-based tools (like the one from
Ivex, for example).

Does this sort of idea appeal to anyone else? IF so, which DSP chip(s)? I
tend to lean towards the ADSP-2181, but then TAPR is sponsoring the DSP56002
development board buy, so that may make more sense to target...

Cheers,

Lyle

3. Re: [HFSIG:1551] Development Environment

From: Jim Sanford <jsanford@infi.net>
Date: Wed, 02 Oct 1996 11:58:19 -0400
To: hfsig@tapr.org
Message-ID: <3252911B.5746@infi.net>
Rendered body type: text/plain

Lyle Johnson wrote:
>
> A couple years ago we bought the development system for the Star
> Semiconductor SPROC chips. OF course, once we had the product designed and
> debugged, Star went belly up and we had to switch horses :-)
>
> The development system they sold consisted of a schematic interface (OrCAD).
> The various building blocks in the schematic library were associatd with
> pre-written and optimized code in both subroutine and in-line formats.
> Arguments needed by the routines were entered onto the schematic as fields
> (much as a resistor could have a "value" and a "tolerance", a signal
> generaotr could have a "frequency" and an "amplitude."
>
> Once you "sketched" your design, you'd tell the schematic capture tools to
> output its "netlist", then turn the results of that loose on the "compiler"
> which would look for syntax problems and, once it was happy, it would chrun
> on the result and provide you with the otuptu code. You'd then download
> this into the development lab and start debugging the design rather than the
> details.
>
> I guess loosely it is like using a high level language rather than assembler
> to get things done.
>
> Of course, the SPORC chip had superb real-time debugging characteristics,
> but that is another story.
>
> I'm wondering if anyone on this sig has the knowledge and inclination to do
> something like this? Perhaps those that are good DSP coders could write the
> various "modules" and someone strong in compiler knowledge could write the
> tool that took the "netlist" and created the source module to run through
> the mfr's assembler/linker/etc.
>
> I could be persuaded to work on schematic symbols and so forth for at least

> one or two popular (inexpensive?) windows-based tools (like the one from
> Ivex, for example).
>
> Does this sort of idea appeal to anyone else? IF so, which DSP chip(s)? I
> tend to lean towards the ADSP-2181, but then TAPR is sponsoring the DSP56002
> development board buy, so that may make more sense to target...
>
> Cheers,
>
> Lyle
Lyle:
I, too, lean toward the ADSP chips and think you have a GREAT idea.
Wish I were smart enough to do it .. . . ..

Good luck1

73, Jim
wb4gcs@amsat.org

4. RE: [HFSIG:1551] Development Environment

From: "Bloom, Jon, KE3Z" <jbloom@arrl.org>
Date: Wed, 2 Oct 1996 14:28:00 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <m0v8V7q-000f4iC@mgate.arrl.org>
Rendered body type: text/plain

Lyle wrote:
> A couple years ago we bought the development system for the Star
> Semiconductor SPROC chips. OF course, once we had the product designed
and
> debugged, Star went belly up and we had to switch horses :-)

That's why they call it the "bleeding edge." :-)

> I'm wondering if anyone on this sig has the knowledge and inclination
to do
> something like this? Perhaps those that are good DSP coders could
write the
> various "modules" and someone strong in compiler knowledge could write
the
> tool that took the "netlist" and created the source module to run
through
> the mfr's assembler/linker/etc.

I've thought about something along these lines from time to time but have
never
had the time to do anything about it--and still don't. (Read: I'm *not*
volunteering.)

> Does this sort of idea appeal to anyone else? IF so, which DSP
chip(s)? I
> tend to lean towards the ADSP-2181, but then TAPR is sponsoring the
DSP56002
> development board buy, so that may make more sense to target...

I think it would at least be worth thinking about doing it in a modular
enough way that it could support different processors by, for example,
loading a processor-specific DLL if it's a Windows app. Doing it that
way would make it more complex initially but much more useful in the
long run.

It also would make sense to have some platform-specific capabilities.
If, for example, you are generating code for a TMS320C26, it would be
nice if the program knew enough about the differences between the
DSP-93 and the TI DSK to insert the needed glue code to generate a
complete loadable program for the target platform.

It also would be neat if the program could do a simulation of the
designed
system, as well as generating processor code. (And since I'm not
volunteering to do the work, it costs me nothing to make the suggestion!)
At first blush, this would seem to be a relatively easy addition to the
scheme.

It seems to me that this could be a pretty straightforward project on the
first cut, particularly if you didn't worry too much about generating
optimal DSP code for the particular target processor, but just strung
together canned chunks of code to implement particular DSP functions
(FIR and IIR filters, for example). Whenever I've thought about this in

the past it was always the graphical front-end that made me shy away
from pursuing it, but I think Lyle has a good idea of a way around that.
 --
Jon Bloom, KE3Z
jbloom@arrl.org

5. RE: [HFSIG:1553] RE: Development Environment

From: Danny Kohn <danny.kohn@systematik.se>
Date: Wed, 2 Oct 1996 21:18:55 +0200
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BBB0AD.07A7F860@dialup99-3-1.swipnet.se>
Rendered body type: text/plain

On den 2 oktober 1996 14:42, Bloom, Jon, KE3Z[SMTP:jbloom@arrl.org] wrote:

> Whenever I've thought about this in
> the past it was always the graphical front-end that made me shy away
> from pursuing it, but I think Lyle has a good idea of a way around that.

Is the ARRL Radio designer not a viable solution to this or is it not
advanced enough?

Hälsningar / Regards
Danny Kohn - SM0NBJ
Utvecklingskonsult / Development Consultant
Systematik Consuliting AB - www.systematik.se - +46 708 140 300

6. Re: [HFSIG:1554] RE: Development Environment

From: Lyle Johnson <lylej@azstarnet.com>
Date: Wed, 2 Oct 1996 18:08:39 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19961002180827.0036ddd0@azstarnet.com>
Rendered body type: text/plain

Danny,

ARD has no graphical front end. You type a textual netlist into it.

Cheers,

Lyle

***

At 03:19 PM 10/2/96 -0500, you wrote:
>On den 2 oktober 1996 14:42, Bloom, Jon, KE3Z[SMTP:jbloom@arrl.org] wrote:
>
>> Whenever I've thought about this in
>> the past it was always the graphical front-end that made me shy away
>> from pursuing it, but I think Lyle has a good idea of a way around that.
>
>Is the ARRL Radio designer not a viable solution to this or is it not
>advanced enough?
>
>Hälsningar / Regards
>Danny Kohn - SM0NBJ
>Utvecklingskonsult / Development Consultant
>Systematik Consuliting AB - www.systematik.se - +46 708 140 300
>
>
>

7. TAPR Office to reduce to minimum operations

From: Greg Jones <wd5ivd@tapr.org>
Date: Thu, 3 Oct 1996 10:01:18 -0500 (CDT)
To: tapr-bb@tapr.org (TAPR-BB mailing), tapr-board@tapr.org (TAPR Board),
        netsig@tapr.org (NETSIG mailing), bbssig@tapr.org (BBS SIG mailing),
        hfsig@tapr.org (HF SIG mailing), dsp-93@tapr.org (DSP-93 Build),
        aprssig@tapr.org (BBS SIG mailing), tapr-sys@tapr.org,
        amsat-bb@amsat.org (AMSAT BB Mail Group)
Message-ID: <199610031501.KAA15057@tapr.org>
Rendered body type: text/plain

TAPR Office to reduce to minimum operations   Oct 3, 1996

The TAPR Office will shutdown to minimum operations starting October 8th and
running at least until October 28th.

This means that Dorothy will not be working the office phones during this
period. The office will continue to operate, but at a much reduced level.
E-mail, US mail, Voice Mail, etc will still be received and recorded, but
everything will be processed at a very much slower rate during this period.

GPS-20 units have begun to ship. As of this e-mail, approximately 40 units
have shipped. We are awaiting additional pins for the connector (suppose to
arrive Oct 11th) in order to complete the remainder of the units. These
pins replace the ones that were incorrect in the shipment we received. Once
the pins arrive, we will get the other units out as soon as possible.

The Motorola EVM56002 have been placed on order and are expected to arrive
sometime after Oct 21st. They will then be shipped out as time permits
under the operational schedule at the office. There are something like 60 of
the 200 units still available in this purchase at the $85 + $10 s/h price.
Check the TAPR Web page for full details www.tapr.org. First come basis on
these units.

TAPR is sorry for any delays that the reduced operations schedule at the
office might cause. If everything turns out positive, the office should be
back operational towards the end of October.

Cheers - Greg Jones, WD5IVD
President, TAPR

(wd5ivd@tapr.org)

8. Re: [HFSIG:1551] Development Environment

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 03 Oct 1996 12:10:04 -0800
To: hfsig@tapr.org
Message-ID: <199610031913.MAA26419@PEAK.ORG>
Rendered body type: text/plain

Hi Lyle,

Nice to see you on HFSIG.

There have been several attempts in doing this kind of thing: The Berkely
Ptolemy project, products from Hyperception, White Mountain and others.
There are also a few toy programs on the web that is interesting.

I cannot make claims or offer any advice on how good they are as I have not
seriously tried any of these myself.

>I'm wondering if anyone on this sig has the knowledge and inclination to do
>something like this? Perhaps those that are good DSP coders could write the
>various "modules" and someone strong in compiler knowledge could write the
>tool that took the "netlist" and created the source module to run through
>the mfr's assembler/linker/etc.
>
>I could be persuaded to work on schematic symbols and so forth for at least
>one or two popular (inexpensive?) windows-based tools (like the one from
>Ivex, for example).
>
>Does this sort of idea appeal to anyone else? IF so, which DSP chip(s)? I
>tend to lean towards the ADSP-2181, but then TAPR is sponsoring the DSP56002
>development board buy, so that may make more sense to target...

I was just wondering; how much of the SPROC effort is still usable and
available for a bootstrap effort? Can we use the Orcad front-end for
example? Perhaps if the format of it's output could be interpreted/compiled
or translated using compiler generator tools such as LEX, or YACC or similar
tools.

Providing the appropiate DSP modules to run the code on a DSP platform may
perhaps need a bit of planning. This kind of modular engineering is perfect
for object-oriented programming design. Code may not quite be as efficient,
but perhaps good enough for serious experimentation on a contemporary DSP.
One of the student papers at the DCC did quite a nice job of showing how
these concepts worked - their case was a satellite ground station
controller, but might just as well have been a communications model.

Not that I'm volunteering either, but perhaps may be interested in a
professional effort. Heck, I was wondering when I was ever going to justify
the time I spent in graduate school learing about FA, FSM's and compiler theory.

My 2 cents worth.

--Johan

9. Eb/No calculations

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 03 Oct 1996 12:10:08 -0800
To: hfsig@tapr.org
Message-ID: <199610031913.MAA26423@PEAK.ORG>
Rendered body type: text/plain

Hi Theoreticians,

I have been playing the numbers game a bit to grasp the significance of Eb/No rather
than power ratios. Tom, N5EG have been kind to help review some of the examples shown here, but if its wrong blame me :-)

It would be greatly appreciated if you could look over these calculations and straighten me out if you spot something amiss.

Comments/critisism or suggestions are welcome - post here for the benifit of others.

Thanks in advance,

--Johan

----------CUT HERE----------

----------NB-(GAUSSIAN CHANNELS ASSUMED)-NB----------
Example 1:

Say we have an AMTOR signal (100 bits/sec, binary FSK), and given that AMTOR throughput goes down to zero at roughly -2.5 dB S/N (4000 Hz bandwidth), what is the equivalent Eb/N0 ?

By definition; "Signal-to-Noise" power ratio = 10log(P/N)
Thus; P/N=antilog(-2.5/10)=0.5623

A few useful relations:
1) N=N0*B where N0=noise density in watts/Hz
2) P=Eb*Rs where Rs=log2(M)/T (M=no of alphabet symbols)
(T=symbol time)
thus P=100*Eb

Then, Eb/(N0*4000)=0.5623 or
Eb/N0=(4000/100)*0.5623=22.492
=10*log(22.492) dB
=13.52 dB

Example 2:

There are claims that the new Pactor 2 protocol can work to a S/N level of -18 dB. Pactor 2 in its most robust mode uses two parallel BPSK carriers, each coded at 100 symbols/sec, which effectively yields 200 bits/sec. What is the equivalent Eb/N0? Would you agree that the claims have some credibility given the Shannon limit of -1.6 dB Eb/N0?

Noise power = N0*3000,
Thus P/N=0.015848

First consider a single carrier;
P=Eb*Rs where Rs=log2(M)/T (M=total no of symbols)
(T=symbol time)
=1/(1/100)
P=100*Eb (normalized, per bit)

The total power for the two carriers is 2*P, however the energy per bit is only half that because there are effectively two bits per symbol.

Thus Eb/N0=(3000/(100))*0.015848= -3.22 dB < -1.6 dB !!

The claims seems marginal, but try a S/N of -16 dB that equals -1.22 dB Eb/N0, which is probably as far down as you can go, in theory at least. The claims are thus within

ball-park numbers, at least, that is how I would interpret it.

Changing the protocol to redundantly replicate bits in the two channels, yields
only half the bit rate, but adds robustness, for then Eb/No at -16 dB S/N
translates 1.78 dB Eb/N0.

Now for comparitive purposes: QUATOR uses four parallel channels at
75 symbols/sec. In the most robust mode, bits are redundantly replicated
over the
four channels. Given this, then; -16 dB S/N corresponds to 6.0 dB Eb/N0.
At this level there should still be throughput (given that the ECC
is doing it's bit) - in fact, it appears that the theoretical limit is
approached around -23 dB S/N, which equals -0.96 dB Eb/N0.

                              Example 3:

QUATOR's modulation scheme uses a symbol rate of 75 baud
and packs 16 bits/symbol under good conditions. Say the system has a
measured BER of 10E^-5 at 11 dB S/N in a 3000 Hz bandwidth. What is
the equivalent Eb/N0?

 Similarly; Noise power = N0*3000,
 Thus   P/N=12.5893

              P=Eb*Rs where Rs=log2(M)/T (M=no of signals)
                                        (T=symbol time)
                           =16/(1/75)
              P=16*75*Eb (normalized, per bit)

 or   Eb/N0=(3000/(16*75))*12.59893=31.4973=14.98 dB

For a few other (much higher error rates, not specified):

               2 dB (3000 Hz BW) equates to   5.98 dB Eb/N0

              -2 dB   "   1.98 dB Eb/N0
(See Example 1, this is the approximate S/N level where AMTOR stops working.
QUATOR, however, seems to be some working margin, it may even be possible to
work
at one of its higher throughput levels - given ECC coding does it's bit.)

              -5 dB   "   -1.02 dB Eb/N0
---

10. Re: [HFSIG:1558] Eb/No calculations

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 3 Oct 1996 15:20:44 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610032220.PAA14157@servo.qualcomm.com>
Rendered body type: text/plain

Johan,

Eb/N0 depends *user data rate*, not symbol rate or signal bandwidth.
Some of your examples appear to use modulation symbol rate in their
calculations, which is incorrect. And you don't mention what coding,
if any, is in use by the three schemes. You can easily get numbers
that appear to beat the Shannon Limit if you look at coded symbol rate
rather than user data rate.

Eb/N0 is derived as follows:

        Eb = S/R
        N0 = N/B
        Eb/N0 = (S/R) / (N/B) = (S/N) * (B/R)

where

S = total rx signal power, watts
N = total rx noise power, watts
B = bandwidth in which N is measured, Hz
R = user data rate, bits/sec (*not* code or modulation symbol rate)

Here's one way to measure Eb/N0.

1. Measure the average signal power. If the modem is a hybrid ARQ type
that relies on code combining (e.g., Pactor), average the power over

some number of transmit-receive cycles. If you want, you can do this
by measuring the TX-on power and multiplying by the transmit duty
cycle.

2. Measure the average user data throughput. Since the data probably
comes out in fast bursts separated by pauses (particularly if the
modem uses FEC and/or ARQ), you can measure the average throughput
across some number of T-R cycles.

3. Divide the average signal power by the average throughput. This
gives the average signal energy per bit, Eb.

4. Measure the noise spectral energy, No. This can be done by
measuring the average received noise power in some known measurement
bandwidth and then dividing by that bandwidth. If you use a spectrum
analyzer, be sure to use a lot of video filtering to flatten out the
"grass" so you measure the true average noise power and not the
momentary fluctuating peak of some narrowband component.

5. Divide Eb by N0.

--Phil

11. Re: [HFSIG:1527] Re: Johan's modem

From: N0AOT@aol.com
Date: Fri, 4 Oct 1996 02:19:38 -0400
To: hfsig@tapr.org, forrerj@peak.org
Message-ID: <961004021937_324734913@emout01.mail.aol.com>
Rendered body type: text/plain

Hi Johan--

What about our beloved DSP56002EVM ??? ;>)

--Bob

In a message dated 96-09-10 21:21:52 EDT, you write:

>>As far as the actual implementation, at least what I am doing; I have been
>>using the PSA sound card platform thus far. This seems to have worked out
>>exteremely well for coding various parts of the modem, i.e., for monitoring
>>the workings of things like Costas loops and bit sync algorithms.
>>For the future, however, I'm not sure which way I'll go. My experience with
>>the new TI 320C31 DSK has been very encouraging - perhaps too little
memory,
>>but it is really nice for the task, the speed and dynamic range, and a high
>>speed connection with the PC. This would make an ideal free-standing
>>implementation. On the other hand, I'm tempted to develop on the PC using
>>the soundblaster, but that would have no chance of ever getting the the ARQ
>>mode to work. I will, however, be able to play with the broadcast and CSMA
>>modes.

12. Re: [HFSIG:1560] Re: Johan's modem

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 04 Oct 1996 08:58:33 -0800
To: hfsig@tapr.org
Message-ID: <199610041601.JAA05817@PEAK.ORG>
Rendered body type: text/plain

Hi Bob,

How are you? Long time.

>Hi Johan--
>
>What about our beloved DSP56002EVM ??? ;>)
>
>--Bob

Don't give up too soon, hopefully I will get to that in the not too distant
future.

The only reason why not the EVM presently, is the lack of a high speed link
between the EVM and the host computer. Experimental work required a detailed
study of various parts of the demodulator such as loop filters and tracking
algorithms working at the sample rate. It is thus not unusual to collect the
internal states over a few bit periods and end up with a huge file on the
host. The soundcard does this elegantly with no sweat. Once things are

working, however, the amount of data will probably be at a much lower rate, i.e., at symbol rates instead of sample rates, also contain only the bare necessities. At this time data would probably be able to be sent over a serial interface. There also still remains a possibility to implement a high speed parallel interface to the EVM for this purpose - you are bound to hear more about this in future.

Have you had the opportunity to do some fun things with your EVM?

73's

--Johan

13. Error Detecting Codes

From: Richard Lay <lay@cod.nosc.mil>
Date: Fri, 4 Oct 96 10:46:52 PDT
To: hfsig@tapr.org
Message-ID: <9610041746.AA28255@marlin.nosc.mil>
Rendered body type: text/plain

I'm interested in detecting errors in a block of bits. I don't really care about correcting them, or even how many errors there are. What options are there?

I've heard of an error detection scheme where you send data blocks encoded so that there are an equal number of ones and zeros. If the number of ones and zeros is not equal on reception, then there is an error. This seems to detect all odd number of error cases, but only about half the even number of error cases. I need to do better than that.

Regards,

Richard Lay

14. Impressions of the 1996 DCC

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 04 Oct 1996 15:28:20 -0800
To: hfsig@tapr.org
Message-ID: <199610042231.PAA20798@PEAK.ORG>
Rendered body type: text/plain

Hi,

I promised that I would give my impressions of the last ARRL Digital Communications Conference (DCC) held in Seattle September 20-22, 1996. There was a short bit that I posted earlier - this is a continuation.

Like most of us, there was only opportunity to attend a few of the activities as much of it was parallel sessions. Also, I did not attend any of the workshops, which was about APRS and high speed networking. My apologies that I probably won't do justice to the work and effort that went into doing these workshops. I heard it was great.

There was the usual series of introductory talks: Digital Communications, HF Digital Communications, and this time; an Introduction to Spread Spectrum (SS). Greg Jones was a very energetic speaker and did a great job on the introductory stuff. Steve Bible gave a thought-provoking introduction to the topic of SS. What was interesting to me was hearing how such a new mode of operation would be like; some form of channelized tuning or scanning operation probably with a very strange-sounding background - like tuning the band but now it was all SS codes. It was evident that SS was on everyone's mind. The room was absolutely packed while the talk next door was only a partially filled up - eventhough that also was a good one.

There were two papers by students participating in the "Best Student's Paper" - that went over well. The quality of their work was excellent. One presentation dealt with the control system for an amateur radio satellite ground station. Talk about sophistication; this one uses the QNX real time operating system (it is small and compact enough that it lives entirely in a Pentuim's L cache) as a basis and designed the various control modules using the object-oriented programming concept called "Actor(s)". I hope that the idea of the Student Awards would continue to produce and attract such talented participants.

Phil Karn presented some of his recent work on concatenated codes. These are combined convolutional and block codes that has some rather neat properties. This talk was outstanding as Phil really is the master of this topic and the content was appreciated by many - judged by the full house. There was an interesting comment on the future of research on coding theory. Phil was talking here about "Turbo codes" and noted that once that have been

formalized, the last bit of coding gain would have been accounted for. So if you were a coding theory theoretician, you would better find another field to work in! However, I suspect there remains a great deal of this facinating theory to be explored in HF digital applications.

Interest in the future development of HF digital was shown in Craig McCartney's (WA8DRZ) talk, which was about a commerial marine radio operation - how HF digital communications and the Internet makes this possible.

The dinner speaker was Lyle Johnson - one of the founding fathers of TAPR. Lyle reminded us that the role and place of Amateur Radio in today's society was never as much in peril as now. Just think of the fact that we are still using 1200 baud packet; that SSB is still the main voice modulation method of choice. How long ago were these technolgies developed? Why have we not made progress? It comes to no surprize that the amateur committment to provide emergency readyness is being challenged by the coming of age of the Internet, robust fiber optic communications, LEO's etc. I hope the gloomy picture will inspire future experimentation and development of our spectrum resources or realize that there are eager eyes wanting to claim it.

The HFSIG meeting was held after dinner and provided intellectual entertainment for those interested in HF. I think this was successful - at least from where I was standing on the other side of the podium! The topic this time was woven around past discussions we have had on HFSIG regarding HF Channel simulation.
Tom McDermott gave an excellent presentation of the mechnics involved in simulation using the Watterson model, that I followed with showing a number of "doppler grams", courtesy of Peter Martinez, G3PLX. These are really unique and worth seeing. A brief overview of software for the DSP-93 and a demo running this implementation of the Watterson ionospheric model, concluded this part of the meeting. I also presented an outline of the work that I have been doing on QUATOR.

I wanted to talk about the future of HFSIG and made a desperate plea for someone to offer to help out. My personal business committments has gotten extremely demanding to the point where I just felt it necessary to step back and let someone else continue with the good work that started, and still continues, on this SIG. Being in the chair position means being at a focal point with a lot happening behind the scenes. One need to process, culture, follow up, and be involved constantly as there are a numerous exiting and worthy opportunities coming your way.

So keep that in mind: Anyone interested to help run HFSIG please get in touch with Greg or me. Otherwise as of the DCC, I will remain an interested party, but probably stay on the sidelines and participate whenever there is an opportunity.

I trust that this summary is of interest - remember that these are my personal views and observations; one person's view of the world - I hope I have most of the facts staight but short memory often let's me down. Thank you much to all that made attending the conference a worthwhile experiece. It was great seeing all and looking forward to the next DCC (which I hear will be out on the East Coast).

73's and take care.

--Johan, KC7WW

15. ARRL & TAPR DCC in the Linux Gazette

From: "Steven R. Bible" <srbible@gnatnet.net>
Date: Fri, 04 Oct 1996 18:36:18 -0400
To: tapr-bb@tapr.org, hfsig@tapr.org, amsat-bb@amsat.org
Message-ID: <1.5.4.32.19961004223618.006b07f0@gnatnet.net>
Rendered body type: text/plain

Phil Hughes, WA6SWR, publisher of the Linux Journal visited the ARRL & TAPR DCC conference in Seattle on September 20-22. I think Phil was pleasently surprised to find that Linux was being used by hams. He's written a short piece in the Linux Gazette, an online magazine about Linux, about the DCC. The article is titled:

    Hams, Packet Radio and Linux

and it's located at:

    http://www.ssc.com/lg/issue10/radio.html

I also have noted that a future issue of the Linux Journal is going to have a piece about satellite tracking. For more info see:

    http://www.ssc.com/

Disclaimer: No I have NOTHING to do with SCC. I'm just a fanatical user of

Linux since version 0.96!

73,

 - Steve, N7HPR
 (n7hpr@tapr.org)

16. Out of Office AutoReply: [HFSIG:1560] HFSIG digest 166

From: Dave Moore <Dave_Moore@ATK.COM>
Date: Fri, 4 Oct 1996 18:04:12 -0500
To: "hfsig@tapr.org" <hfsig@tapr.org>
Message-ID: <c=US%a=_%p=ATK%l=EXCHANGE_MN1-961004230412Z-29007@Exchange_MN1.ATK.COM>
Rendered body type: text/plain

Out until Monday 10/7, but will check VoiceMail and check mail in the
evenings. May be reached at 921-8844 for more urgent messages.
Thanks, Dave Moore

17. EVM56002 Group Purchase Update

From: Greg Jones <wd5ivd@tapr.org>
Date: Sat, 5 Oct 1996 02:53:54 -0500 (CDT)
To: tapr-bb@tapr.org (TAPR-BB mailing),
        amsat-bb@amsat.org (AMSAT BB Mail Group), dsp-93@tapr.org,
        hfsig@tapr.org (HF SIG mailing)
Message-ID: <199610050753.CAA14873@tapr.org>
Rendered body type: text/plain

TAPR EVM56002 Group Purchase Update

The last sixty TAPR EVM56002 group purchase units have been spoken for.
Purchase is closed.

Cheers - Greg, WD5IVD

18. ARRL DCC

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Sat, 5 Oct 1996 19:30:41 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <199610060230.AA008609042@polecat.sr.hp.com>
Rendered body type: text/plain

forrerj@peak.org (Johan Forrer) wrote:

>I promised that I would give my impressions of the last ARRL Digital
>Communications Conference (DCC) held in Seattle September 20-22, 1996.

An excellent summary for those of us who couldn't go. I wish I had time
to attend more of these gatherings. Too bad it wasn't the next weekend
so I could have combined it with a business trip to Spokane that Monday.

>   The dinner speaker was Lyle Johnson - one of the founding fathers of
>TAPR. Lyle reminded us that the role and place of Amateur Radio in today's
>society was never as much in peril as now. Just think of the fact that we
>are still using 1200 baud packet; that SSB is still the main voice
>modulation method of choice. How long ago were these technolgies developed?
>Why have we not made progress?

It's interesting to note that SSB is a very spectrum-efficient mode,
even by today's standards. You can pack about 10 SSB signals (more
using spectrum-folding techniques) in the space of one analog cellular
channel, which is better than most of the digital modes in use today.
Don't get me wrong -- there are lots of good reasons to go digital.
(Cost, size, reliability, easier integration of modems and other new
features.) But the oft-heard statement that "digital modes are much
more spectrum-efficient than analog" is only true if you compare modern
digital technology with the 50-year-old wideband FM currently used
on the US cellular bands.

Alan Bloom N1AL

19. Re: [HFSIG:1567] ARRL DCC

From: Lyle Johnson <lylej@azstarnet.com>
Date: Sat, 5 Oct 1996 21:49:28 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19961005214925.003584a8@azstarnet.com>
Rendered body type: text/plain

Alan,

Actually, as I understand things, for a given amount of power and Hz, you
can send more (voice) information using SS techniques than you could using SSB.

I remember making long distance phone calls a decade or two ago, and you
often heard the FDM SSB chatter on adjacent channels on the overseas
circuit. Now, telephone systems are digital, and I would have to guess that
if the satellite transponders (a definite power limited situation) could
squeeze more voice cirvuits in using an FDM SSB system than they can over
the same path using digital techniques, they would.

SOm eof the more astute spread spectrum guys can probably give us alot of
informaitonm on how a given system can actually handle more voices using
these technqieus than can be done using SSB.

I think the problem is that spread spectrum techniques are "non intuitive" -
you just somehow *know* deep inside that narrowband is how you slice things
finer :-)

Having said that, I ahve certainly enjoyed using SSB over the years, and it
is lots better than many other methods!

Cheers,

Lyle

>It's interesting to note that SSB is a very spectrum-efficient mode,
>even by today's standards. You can pack about 10 SSB signals (more
>using spectrum-folding techniques) in the space of one analog cellular
>channel, which is better than most of the digital modes in use today.
>Don't get me wrong -- there are lots of good reasons to go digital.
>(Cost, size, reliability, easier integration of modems and other new
>features.) But the oft-heard statement that "digital modes are much
>more spectrum-efficient than analog" is only true if you compare modern
>digital technology with the 50-year-old wideband FM currently used
>on the US cellular bands.

20. Re: [HFSIG:1557] Re: Development Environment

From: Lyle Johnson <lylej@azstarnet.com>
Date: Sat, 5 Oct 1996 22:04:09 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19961005220406.0036a760@azstarnet.com>
Rendered body type: text/plain

Johann,

>There have been several attempts in doing this kind of thing: The Berkely
>Ptolemy project, products from Hyperception, White Mountain and others.
>There are also a few toy programs on the web that is interesting.

The project I was proposing was to create an environment that could be used
on a standalone DSP to operate as a real-time system. I know that some of
these you mention work with a plug-in card (usually big buck$).

For example, at work we evaluated a communications system from "IQ Comm" in
New York. This is an $8,000 software package that lets you model
communications channels, etc. (we sent it back). The problem is that they
have nice little building blocks that implmemnt "perfect" QPSK demodulators,
for example, but give no clue as to how to implmement such a demodualtor,
much less implmenment it for a given processor in real time.

The SPROC system, on the other hand, didn't model anything. Essentially,
they had pre-compiled a large number of functional blocks (AGC, sine
generators of specified accuracy, "VCOs," ampliers, mixers (real and
complex)) and so forth. The schematic capture system was used as a
graphical front end to tie these blocks together and embed the needed
parameters to properly operate (e.g., sampling rate, gain, attach and decay
rates, and so forth). They also kindly included source code for all
functions and it was heavily commnted (and copyrighted). Now that they are
and have been defunct for quite some time, maybe the copyright isn't a big
deal? They had a 24-bit processor that used a 2.22 number format, so it was
a little odd, but the Motorola EVM could probably handle it easily enough.

>...
>I was just wondering; how much of the SPROC effort is still usable and
>available for a bootstrap effort? Can we use the Orcad front-end for
>example? Perhaps if the format of it's output could be interpreted/compiled
>or translated using compiler generator tools such as LEX, or YACC or similar

>tools.

Orcad can generate a number of netlist formats. Hoever, Ivex has a windows based schematic package that also support various netlist formats (I beleive) and is really cheap - free download with 100 "pins" (signals in our case - ought to be plenty) and for $29.95 they increase that to 250 "pins." For a few hundred $ they have unlimited size, but for this application we can get by with either the free of the $30 version, I suspect. Check out "http://www.ivex.com" right there in Oregon - excellent package IMHO.

>Providing the appropiate DSP modules to run the code on a DSP platform may
>perhaps need a bit of planning. This kind of modular engineering is perfect
>for object-oriented programming design. Code may not quite be as efficient,
>but perhaps good enough for serious experimentation on a contemporary DSP.
>One of the student papers at the DCC did quite a nice job of showing how
>these concepts worked - their case was a satellite ground station
>controller, but might just as well have been a communications model.

I think you are correct here. We'd need to identify the modules we want to start with (enough to make some interesting implmenetations; not too many to discourage the effort from ever taking off) and then apportion some tasks. Once we got it running for one DSP, it might not be that hard to make it work for others... ?

>Not that I'm volunteering either, but perhaps may be interested in a
>professional effort. Heck, I was wondering when I was ever going to justify
>the time I spent in graduate school learing about FA, FSM's and compiler
theory.

No one is volunteering :-(but that doesn't mean we can't do it anyway :-)

I think the debug environment is key to making this worthwhile. SPROC really was an outstanding chip in this regard, since you could look at anyting in real time and alter memory locations on the fly. SOme of the DSP chips with JTAG or JTAG-like serial pprts allow similar things (I'm thinking particularly of the TMS320C31 and the newer Motorola devices - maybe the DMA interface on the ADSP-2181 woudl also allow on-the-fly inspection and modification).

Food for thought.

Cheers,

Lyle

21. Unsubscribe

From: Louis Dupree <cn1743@coastalnet.com>
Date: Sun, 06 Oct 1996 13:56:42 -0400
To: hfsig@tapr.org
Message-ID: <2.2.32.19961006175642.006c0aa4@abaco.coastalnet.com>
Rendered body type: text/plain

How do I unsubscribe from this group? I have tried and says I am not on the list. I am getting emil for this group. I enjoy it, but I am going to be away for several weeks, and I do not want to miss my personal email and my ISP willonly store so much. Thank you. 73's Louis
Louis J. Dupree Jr.
3015 Englewood Dr.
Kinston, NC 28504
ldupree@coastalnet.com

22. Re: [HFSIG:1569] Re: Development Environment

From: forrerj@peak.org (Johan Forrer)
Date: Sun, 06 Oct 1996 11:31:46 -0800
To: hfsig@tapr.org
Message-ID: <199610061834.LAA04836@PEAK.ORG>
Rendered body type: text/plain

Hi Lyle,

> The problem is that they
>have nice little building blocks that implmemnt "perfect" QPSK demodulators,
>for example, but give no clue as to how to implmement such a demodualtor,
>much less implmenment it for a given processor in real time.

I suspect that some of these are attempts to generalize the world and thus end up being as big and unwieldy as they are. Perhaps a more specific solution is what is needed.

>
>The SPROC system, on the other hand, didn't model anything. Essentially,
>they had pre-compiled a large number of functional blocks (AGC, sine
>generators of specified accuracy, "VCOs," ampliers, mixers (real and
>complex)) and so forth. The schematic capture system was used as a
>graphical front end to tie these blocks together and embed the needed
>parameters to properly operate (e.g., sampling rate, gain, attach and decay
>rates, and so forth).

.....

> Hoever, Ivex has a windows
>based schematic package that also support various netlist formats (I
>beleive) and is really cheap - free download with 100 "pins" (signals in our
>case - ought to be plenty) and for $29.95 they increase that to 250 "pins."
>For a few hundred $ they have unlimited size, but for this application we
>can get by with either the free of the $30 version, I suspect. Check out
>"http://www.ivex.com" right there in Oregon - excellent package IMHO.
>

I'll have a look at this.

Without getting into copyright issues, would it be possible to compile a
list of the basic functional blocks, a brief functional description, and a
list of their respective inputs and outputs? (I'll be willing to do this if
the materials can be made available.)

Presently we use software with the 56002 EVM that is based on the "Leonid"
kernel. Using these kernel services, relieves the code developer from a
number of tricky tasks such as dealing with low-level code for the CODEC's
real time data streams, serial communications etc. These services are
implemented by using assembly-language macro calls. It would certainly be
possible to build and extend a library of macros to include these functional
blocks as described above.

Conceiveably then, an assembly language programmer can then put a working
commmunications model together simply by including the appropiate macros.
Having this macro library available then, it would perhaps be a little
easier to build a bridge between the model description language.

How does this sound?

--Johan

23. Turbo Codes

From: Charles Brain <chbrain@dircon.co.uk>
Date: Sun, 06 Oct 1996 19:23:32 +0100
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19961006182332.0066333c@popmail.dircon.co.uk>
Rendered body type: text/plain

Does anybody know of any good links to information on Turbo Codes.

Thanks

Charles (G4GUO)

24. ARRL DCC

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Sun, 6 Oct 1996 12:52:23 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <199610061952.AA008811543@polecat.sr.hp.com>
Rendered body type: text/plain

Lyle Johnson <lylej@azstarnet.com> wrote:

>Actually, as I understand things, for a given amount of power and Hz, you
>can send more (voice) information using SS techniques than you could using SSB.

Of course, the answer you get depends on the assumptions you make, but
in general that is not true. Compared with other digital modes, the
"coding gain" you get with spread spectrum is cancelled out by the extra
noise from the greater bandwidth. And an SSB signal is actually more
efficient than uncoded digital modulation. For example, if the sampling
frequency is 10 kHz, with 10 bits per sample, the data rate would be
100 kbps, or over 100 kHz bandwidth for GMSK. The SSB signal's 3 kHz
bandwidth has an inherent 15 dB advantage in signal/noise ratio.

For a digital system to get power/bandwidth efficiency similar to SSB,

it must use speech coding to get the data rate down. More complex
modulation modes reduce the bandwidth at the expense of RF S/N ratio
required. And channel coding improves the received S/N ratio, assuming
a minimum RF signal level. Of course, there are also things you can do
with SSB signals to improve the bandwidth and S/N, like spectrum folding
and speech companding. I suspect that SSB is still very competitive for
applications where a low received S/N ratio is acceptable, like amateur
HF DX'ing, but that state-of-the-art digital modes are superior for
applications where you need telephone-quality audio, like cellular radio.

>Now, telephone systems are digital, and I would have to guess that
>if the satellite transponders (a definite power limited situation) could
>squeeze more voice cirvuits in using an FDM SSB system than they can over
>the same path using digital techniques, they would.

Not necessarily. Customers demand high-quality audio, where digital
modes have an advantage. Also all-digital systems are cheaper, easier
to maintain and more convenient for sending digital data.

As I said before, there are good reasons for going digital, but "digital
modulation is more spectrum efficient than analog" is not one of them.

Alan Bloom N1AL

25. [HFSIG:1570] Unsubscribe

From: Louis Dupree <cn1743@coastalnet.com> (by way of Louis Dupree <cn1743@coastalnet.com>)
Date: Sun, 06 Oct 1996 17:43:25 -0400
To: hfsig@tapr.org
Message-ID: <2.2.32.19961006214325.00678168@abaco.coastalnet.com>
Rendered body type: text/plain

How do I unsubscribe from this group? I have tried and says I am not on the
list. I am getting emil for this group. I enjoy it, but I am going to be
away for several weeks, and I do not want to miss my personal email and my
ISP willonly store so much. Thank you. 73's Louis
Louis J. Dupree Jr.
3015 Englewood Dr.
Kinston, NC 28504
ldupree@coastalnet.com

26. Re: [HFSIG:1571] Re: Development Environment

From: Lyle Johnson <lylej@azstarnet.com>
Date: Sun, 6 Oct 1996 18:05:52 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19961006180550.00372be8@azstarnet.com>
Rendered body type: text/plain

Johan,

It sounds good to me. It'll provide a great test of the concept, I think.

I'll dig up my old SPROC literature and list out the modules they included
(it was a big list, so be patient...)

Cheers,

Lyle

***
>Without getting into copyright issues, would it be possible to compile a
>list of the basic functional blocks, a brief functional description, and a
>list of their respective inputs and outputs? (I'll be willing to do this if
>the materials can be made available.)
>
>Presently we use software with the 56002 EVM that is based on the "Leonid"
>kernel. Using these kernel services, relieves the code developer from a
>number of tricky tasks such as dealing with low-level code for the CODEC's
>real time data streams, serial communications etc. These services are
>implemented by using assembly-language macro calls. It would certainly be
>possible to build and extend a library of macros to include these functional
>blocks as described above.
>
>Conceiveably then, an assembly language programmer can then put a working
>commmunications model together simply by including the appropiate macros.
>Having this macro library available then, it would perhaps be a little
>easier to build a bridge between the model description language.
>
>How does this sound?

27. Re: [HFSIG:1570] Unsubscribe

From: forrerj@peak.org (Johan Forrer)
Date: Sun, 06 Oct 1996 21:12:15 -0800
To: hfsig@tapr.org
Message-ID: <199610070414.VAA26234@PEAK.ORG>
Rendered body type: text/plain

>How do I unsubscribe from this group? I have tried and says I am not on the
>list. I am getting emil for this group. I enjoy it, but I am going to be
>away for several weeks, and I do not want to miss my personal email and my
>ISP willonly store so much. Thank you. 73's Louis
>Louis J. Dupree Jr.
>3015 Englewood Dr.
>Kinston, NC 28504
>ldupree@coastalnet.com
>
>

An e-mail message to listserv@tapr.org with UNSUBSCRIBE HFSIG in the body
will do it. Otherwise, use your web browser and it will guide you to
unsubscribe from any SIG.

Hope it helps.

Johan

28. Re: [HFSIG:1572] Turbo Codes

From: Rolf Sommerhalder <rs@ife.ee.ethz.ch>
Date: Mon, 7 Oct 1996 07:58:57 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199610070558.HAA08461@gillespie.ee.ethz.ch>
Rendered body type: text/plain

> Does anybody know of any good links to information on Turbo Codes.
> Charles (G4GUO)

Try
http://class1.ee.virginia.edu/CSL/turbo_codes/
and
http://charli.Levels.UniSA.Edu.Au:80/~steven/turbo/

73,
Rolf, HB9CWP

29. Re: [HFSIG:1562] Error Detecting Codes

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 7 Oct 1996 01:49:54 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610070849.BAA21733@servo.qualcomm.com>
Rendered body type: text/plain

>I'm interested in detecting errors in a block of bits. I don't really care
>about correcting them, or even how many errors there are. What options are
>there?

Any block code can detect errors with relatively little overhead. It's
only if you want to *correct* a lot of them that you start needing a
lot of overhead. Have you considered a simple CRC?

>I've heard of an error detection scheme where you send data blocks encoded
>so that there are an equal number of ones and zeros. If the number of ones
>and zeros is not equal on reception, then there is an error. This seems to
>detect all odd number of error cases, but only about half the even number of
>error cases. I need to do better than that.

That's not a very strong code for error detection purposes; in coding
speak, the "minimum distance" is only 2, which is very small (ie, it
is easily fooled by errors). This kind of code is much more useful as
a way to control the signal spectrum, as keeping the numbers of 0s and
1s equal tends to get rid of low frequency components. Again, just try
a CRC.

Phil

30. Re: [HFSIG:1567] ARRL DCC

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 7 Oct 1996 03:10:02 -0700 (PDT)
To: hfsig@tapr.org

Message-ID: <199610071010.DAA21955@servo.qualcomm.com>
Rendered body type: text/plain

>It's interesting to note that SSB is a very spectrum-efficient mode,
>even by today's standards. You can pack about 10 SSB signals (more
>using spectrum-folding techniques) in the space of one analog cellular
>channel, which is better than most of the digital modes in use today.
>Don't get me wrong -- there are lots of good reasons to go digital.
>(Cost, size, reliability, easier integration of modems and other new
>features.) But the oft-heard statement that "digital modes are much
>more spectrum-efficient than analog" is only true if you compare modern
>digital technology with the 50-year-old wideband FM currently used
>on the US cellular bands.

Warning: hot button alert!

The proper measure of spectrum efficiency in a cellular environment is
not QSOs per megahertz, but QSOs per megahertz PER SQUARE KILOMETER.

There's a very good reason (other than frequency stability and gain
control reasons) why AMPS (analog) cellular uses FM instead of
SSB. Thanks to its capture effect, FM can provide telephone quality
audio with about 10-11 dB SNIR (signal-to-noise-plus-interference), as
compared to about 25-30 dB for SSB.

Sure, SSB channels would have been narrower. But each channel would
have been usable in a *much* smaller fraction of the cells due to the
greater sensitivity to co-channel interference. This would more than
squander the "savings" due to the narrower channel.

Even FM AMPS generally uses each 30 KHz channel in only 1/7 of the
cells, which is a pretty stiff penalty. CDMA (spread spectrum) cellular
systems can use the same channel in *every* cell, making for a much
greater (10-15x) overall system capacity for the same cell sites and
spectral allocations.

Only now is it becoming generally understood that improving spectral
efficiency in an interference-limited environment means going *wider*,
not narrower. Unfortunately not even the FCC is immune to spurious
claims of spectral "efficiency" from narrowband systems -- we lost the
220-222 MHz band to one such claim.

Phil

31. Re: [HFSIG:1571] Re: Development Environment

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 7 Oct 1996 03:17:52 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610071017.DAA21965@servo.qualcomm.com>
Rendered body type: text/plain

> The problem is that they
>have nice little building blocks that implmemnt "perfect" QPSK demodulators,
>for example, but give no clue as to how to implmement such a demodualtor,
>much less implmenment it for a given processor in real time.

Speaking from now considerable personal experience, I can say that
implementing a perfect QPSK demodulator in DSP is trivial. It's just a
pair of multiplies -- no big deal.

The real trick is in deriving the incoming carrier phase reference.
And doing this well is a real bitch (if not downright impossible) at
low data rates on channels with the slightest bit of unpredictable
movement (satellite or ionospheric motion, etc).

I now understand why QPSK is generally reserved for high speed channels
like DSS downlinks at 40 megabits/sec.

Phil

32. Re: [HFSIG:1573] ARRL DCC

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 7 Oct 1996 03:33:47 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610071033.DAA22002@servo.qualcomm.com>
Rendered body type: text/plain

>Of course, the answer you get depends on the assumptions you make, but
>in general that is not true. Compared with other digital modes, the

It *is* true if you consider IS-95 CDMA. See my web page for some
pointers to various papers.

http://www.qualcomm.com/people/pkarn/cdma.html

>in general that is not true. Compared with other digital modes, the
>"coding gain" you get with spread spectrum is cancelled out by the extra
>noise from the greater bandwidth. And an SSB signal is actually more

Eh? It's true that spreading with direct sequence or frequency hopping
by itself does not change the Eb/N0 requirement, i.e., it's
"power-neutral". But if you make meaningful FEC part of the bandwidth
expansion, you come out well ahead. (Yes, you can think of FEC as a
form of spread spectrum -- after all, it *does* make the signal wider
than it otherwise would be if it were uncoded, right?)

>efficient than uncoded digital modulation. For example, if the sampling
>frequency is 10 kHz, with 10 bits per sample, the data rate would be
>100 kbps, or over 100 kHz bandwidth for GMSK. The SSB signal's 3 kHz
>bandwidth has an inherent 15 dB advantage in signal/noise ratio.

A red herring. Nobody sends uncompressed digital voice over mobile radio.

>For a digital system to get power/bandwidth efficiency similar to SSB,
>it must use speech coding to get the data rate down. More complex
>modulation modes reduce the bandwidth at the expense of RF S/N ratio
>required. And channel coding improves the received S/N ratio, assuming
>a minimum RF signal level. Of course, there are also things you can do

Yes, of course -- all digital cellular systems use speech coding. But
by "more complex" modulation modes I assume you mean higher numbers of
bits per symbol, which is exactly the *wrong* thing to do for spectral
efficiency as that necessarily increases the Eb/N0 requirements.
Minimizing the required Eb/N0 is the key to an efficient digital radio
system, and that necessarily requires wider bandwidths.

>a minimum RF signal level. Of course, there are also things you can do
>with SSB signals to improve the bandwidth and S/N, like spectrum folding
>and speech companding. I suspect that SSB is still very competitive for
>applications where a low received S/N ratio is acceptable, like amateur
>HF DX'ing, but that state-of-the-art digital modes are superior for
>applications where you need telephone-quality audio, like cellular radio.

Information theory still applies, whether the signal is analog or
digital. Anything you do to reduce bandwidth (without removing
information) is going to require extra power and make it harder to
reuse the channel on a geographical basis. That's unavoidable.

You're right that SSB works reasonably well at very low S/N ratios,
but I don't consider low SNRs acceptable even on ham radio. Who says
we shouldn't strive for decent audio quality, especially now that it's
within our reach? It's comparing apples and oranges to claim that SSB
"outperforms" digital modes when the quality setpoints are so
different.

Phil

33. Re: [HFSIG:1580] Re: Development Environment

From: Lyle Johnson <lylej@azstarnet.com>
Date: Mon, 7 Oct 1996 07:27:36 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19961007072734.0036c944@pop.azstarnet.com>
Rendered body type: text/plain

>The real trick is in deriving the incoming carrier phase reference.
>And doing this well is a real bitch (if not downright impossible) at
>low data rates on channels with the slightest bit of unpredictable
>movement (satellite or ionospheric motion, etc).

Yes, precisely. THe "block" on the "simulator" just used the clock from the
transmit side and didn't bother to derive *any* clocks. It was OK for
evaluating some things, but not nearly as useful as its price tag would have
indicated...

Cheers,

Lyle

34. Async 4800Bd via FSK Tekk-Radio

From: Ulf.Kumm@t-online.de (Ulf Kumm)
Date: Mon, 7 Oct 96 17:53 MET DST
To: hfsig@tapr.org
Message-ID: <m0vAHzh-0008TtC@mailto00.btx.dtag.de>
Rendered body type: text/plain

I have to transmit a asynchronous (RS232) signal (4800 Baud) via a radio link.
The link should be fully transparent and must not have the delay of a
packet-type link.
The TEKK-radio KS960 is capable to transmit/receive up to 9600Bd FSK,
AF-bandwidth is about 100 to 5000Hz. The link should transmit real time DGPS
data.

* Just connecting RS232 with the FSK-modulator-input does not work: the
floating decider
  needs some transitions to keep his trigger in the middle between 0 and 1,
but will
  loose sync while the RS232 signal pauses for some seconds.
* Any synchronous methods (manchester, hdlc or other bit-stuffing stuff) are
difficult
  because RS232 is asynchronous. But a 1 Byte delay is acceptable.
* I have to generate some edges on the FSK-signal to keep the RX-decider
working while
  the RS232 input pauses.

The problem is easy to solve when the baudrate is just 1200 or 2400, I used
a simple
AFSK chip (TCM3105 or even XR2211/06). But these chips may not run with 4k8
(even if
they do, the bandwidth would be >5kHz). The problem is, that my
FSK-transceivers are
AC-coupled and cannot be used with AF-frequencies below maybe 10 Hz, but
RS232 is a
really static DC-signal.

Does anybody have a simple solution to my problem ?
(I know, that there may exist a lot of complicated ones)
Thanks for your hints!

Ulf Kumm, DK9SJ@DB0RBS.#BW.DEU.EU

Ulf Kumm, DK9SJ,   D-70597 Stuttgart, Johannes-Kraemer-Str. 34
eMail: Ulf.Kumm@t-online.de   Tel: +49 711 7678-925 Fax: -924
----

35. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 11:54:15 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071554.AA00083@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 09:54:15 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

36. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 11:54:11 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071554.AA00037@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 09:54:10 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

37. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 11:54:52 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071554.AA00451@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 09:54:52 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

38. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 11:55:09 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071555.AA00587@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 09:55:09 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

39. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 11:55:00 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071555.AA00513@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 09:55:00 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

40. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 11:55:07 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071555.AA00567@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 09:55:07 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

41. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 11:54:27 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071554.AA00204@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 09:54:27 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

42. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 11:54:46 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071554.AA00395@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 09:54:46 1996

Execution request failed:

```
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig
```

43. Execution failed

```
From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 11:54:34 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071554.AA00290@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct  7 09:54:34 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig
```

44. Execution failed

```
From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 11:54:18 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071554.AA00111@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct  7 09:54:18 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig
```

45. Execution failed

```
From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 11:54:20 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071554.AA00134@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct  7 09:54:20 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig
```

46. Execution failed

```
From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 11:55:24 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071555.AA00698@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct  7 09:55:24 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig
```

47. Execution failed

```
From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 11:55:11 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071555.AA00607@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct  7 09:55:11 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig
```

48. DSP filtering project

```
From: RLANIER@mailb.harris.com (RLANIER)
Date: Mon, 7 Oct 1996 12:56:39 -0400
```

To: hfsig@tapr.org
Message-ID: <25936160@mailb.harris.com>
Rendered body type: text/plain

I am currently working on a "simple" DSP filtering project for a
portable AM/FM radio. I am taking the guts from an old cassette-radio
I had and putting it into a new case along with a DSP chip (I'm using
the ADSP-2101). My idea is to filter out the noise generated by the
environment in an office (lights, computers, etc.) so I can listen to
talk radio :) . The frequencies will be on the AM band; FM is not
affected. As you know, AM is greatly affected by noise. Building the
DSP radio is not a problem, its the filtering part that is in
question. Someone told me that filtering will not help in this
situation. Is this true?

If you have any opinions, helpful hints or good advice, I would
appreciate hearing from you.

73s de
Tony, KE4ATO

49. Re: [HFSIG:1594] Execution failed

From: Greg Jones <wd5ivd@tapr.org>
Date: Mon, 7 Oct 1996 12:26:24 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199610071726.MAA07588@tapr.org>
Rendered body type: text/plain

The group might get a few of these things -- but we have removed the account
in question.

Just press DELETE :-)

Greg

>
> Message from UUCP on combdyn Mon Oct 7 09:54:20 1996
>
> Execution request failed:
>   rmail combdyn.com!hfsig
> Standard error output was:
> rmail: Return to uunet.ca!tapr.org!hfsig
>
>

50. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 13:27:27 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071727.AA01401@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 11:27:27 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

51. UUCP Help

From: Tim Baggett <tim@NMSU.Edu>
Date: Mon, 7 Oct 1996 12:33:29 -0600 (MDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.91.961007122936.6652C-100000@wilma>
Rendered body type: text/plain

HELP!!! Will someone *PLEASE* make _it_ stop!

73
Tim, AA5DF

(Sorry, just a nervous twitch created by the incoming UUCP Errors as I'm
trying to wade through a few hundred emails piled up while away) Reminds
me of watching a BBS forward ;-)

52. Re: [HFSIG:1597] DSP filtering project

From: Glen Worstell <Glen_Worstell@notes.seagate.com>
Date: 7 Oct 96 13:58:46
To: RLANIER <RLANIER@mailb.harris.com>
Cc: hfsig <hfsig@tapr.org>
Message-ID: <9610071901.AA7975@notes.seagate.com>
Rendered body type: text/plain

>... My idea is to filter out the noise generated by the
environment in an office (lights, computers, etc.) so I can listen to
talk radio :) ....Someone told me that filtering will not help in this
situation. Is this true?

Depends on the nature of the noise, but in general filtering can help
quite a bit. Modern ham radio receivers have a noise blanker, and
with impulse noise of various sorts it can make a big difference.

Usually the signal needed to generate the noise-reduction comes
before the narrow band IF.

cheers,
glen.

53. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 13:43:03 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071743.AA01512@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 11:43:03 1996

Execution request failed:
rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

54. (no subject)

From: (unknown sender)
Date: Mon, 7 Oct 1996 13:43:06 -0400
Rendered body type: text/plain

From: combdyn!uucp@uunet.ca (UUCP Admin)
Posted-Date: Mon, 7 Oct 96 11:43:06 -0600
Message-Id: <9610071743.AA01532@combdyn>
Subject: Execution failed
To: tapr.org!hfsig@uunet.ca

Message from UUCP on combdyn Mon Oct 7 11:43:05 1996

Execution request failed:
rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

55. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 13:43:12 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071743.AA01592@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 11:43:12 1996

Execution request failed:
rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

56. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 13:43:16 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071743.AA01632@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 11:43:16 1996

Execution request failed:
rmail combdyn.com!hfsig

Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

57. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 13:43:18 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071743.AA01652@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 11:43:18 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

58. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 13:43:10 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071743.AA01572@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 11:43:10 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

59. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 13:43:08 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071743.AA01552@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 11:43:08 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

60. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 13:43:14 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071743.AA01612@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 11:43:14 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

61. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 15:29:48 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071929.AA02216@combdyn>
Rendered body type: text/plain

Message from UUCP on combdyn Mon Oct 7 13:29:47 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig

62. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 15:29:51 -0400
To: tapr.org!hfsig@uunet.ca

Message-ID: <9610071929.AA02238@combdyn>
Rendered body type: text/plain

```
Message from UUCP on combdyn Mon Oct  7 13:29:51 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig
```

63. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 15:29:54 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071929.AA02258@combdyn>
Rendered body type: text/plain

```
Message from UUCP on combdyn Mon Oct  7 13:29:54 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig
```

64. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 15:29:58 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071929.AA02278@combdyn>
Rendered body type: text/plain

```
Message from UUCP on combdyn Mon Oct  7 13:29:57 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig
```

65. Execution failed

From: combdyn!uucp@uunet.ca (UUCP Admin)
Date: Mon, 7 Oct 1996 15:30:03 -0400
To: tapr.org!hfsig@uunet.ca
Message-ID: <9610071930.AA02314@combdyn>
Rendered body type: text/plain

```
Message from UUCP on combdyn Mon Oct  7 13:30:03 1996

Execution request failed:
        rmail combdyn.com!hfsig
Standard error output was:
rmail: Return to uunet.ca!tapr.org!hfsig
```

66. WWVB to QRO

From: Mike Fahmie <MPFahmie@lbl.gov>
Date: Mon, 07 Oct 1996 13:59:05 -0700
To: hfsig@tapr.org
Message-ID: <199610072102.OAA29719@mh1.lbl.gov>
Rendered body type: text/plain

```
According to an article in Wireless Systems Design (Sept,96), the NIST 60
KHz Time & Frequency station, WWVB will be increasing power.  The new
transmitter should be 'on line' by September of 1997 and will output over 40
KW, at least a 6 dB increase over the present system.  According to the
article, power is being increased so that smaller receive antennas may be used.

Anyone know if MSF is still running 50 KW from Rugby, England?  If so,
WWVB's QRO may cause some headaches for fringe MSF users.  I drove past the
Rugby site in 1989, biggest antenna farm I've ever seen!

-Mike-

                Mike Fahmie   -   Advanced Light Source
                Lawrence Berkeley National Laboratory
                (510) 486-4030     -     MPFahmie@lbl.gov
                                   WA6ZTY @ N6EEG
```

67. Re: [HFSIG:1582] Re: Development Environment

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 7 Oct 1996 15:45:16 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610072245.PAA23433@servo.qualcomm.com>
Rendered body type: text/plain

>Yes, precisely. THe "block" on the "simulator" just used the clock from the
>transmit side and didn't bother to derive *any* clocks. It was OK for

This is *cheating*! Anybody can do this and get great results. Doing it
on a real channel, with all its phase perturbations, is the real trick.

To the extent you can predict sources of phase variations and subtract
them out (e.g., by running a satellite tracking program to preduct
doppler) you can make things work better. But there are sources of
doppler that will surprise you. For example, James Miller pointed out
to me that the S-band antenna on AO-13 is not mounted on the spin axis
of the spacecraft, so when the spacecraft is off pointed there is a
sinusoidal velocity component on the antenna as the spacecraft
rotates. This can amount to 5 Hz. This is very significant with low
rate QPSK!

Even 3-axis-stabilized geostationary satellites are not immune. Though
there may be no doppler shift from the physical motion of the
satellite, there's still ionospheric scintillation. This can be
surprisingly great even at microwave frequencies when the sun is
active.

Phil

68. Re: [HFSIG:1577] Re: Turbo Codes

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 07 Oct 1996 16:52:22 -0800
To: hfsig@tapr.org
Message-ID: <199610072354.QAA01568@PEAK.ORG>
Rendered body type: text/plain

>> Does anybody know of any good links to information on Turbo Codes.
>> Charles (G4GUO)
>
>Try
>http://class1.ee.virginia.edu/CSL/turbo_codes/
>and
>http://charli.Levels.UniSA.Edu.Au:80/~steven/turbo/
>
>73,
>Rolf, HB9CWP
>
>

Thanks Rolf!

--Johan,KC7WW

69. Re: [HFSIG:1615] WWVB to QRO

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 7 Oct 1996 18:28:48 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610080128.SAA23784@servo.qualcomm.com>
Rendered body type: text/plain

>According to an article in Wireless Systems Design (Sept,96), the NIST 60
>KHz Time & Frequency station, WWVB will be increasing power. The new

Interesting. With the GPS revolution I would have thought that the
market for WWVB time & frequency receivers would be on the way
out. Not only are the GPS receivers cheaper, but it's a lot easier to
find roof space for their antennas.

Phil

70. Re: [HFSIG:1615] WWVB to QRO

From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 08 Oct 1996 06:23:33 +0100
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19961008052333.0067036c@popmail.dircon.co.uk>
Rendered body type: text/plain

>Anyone know if MSF is still running 50 KW from Rugby, England? If so,
>WWVB's QRO may cause some headaches for fringe MSF users. I drove past the
>Rugby site in 1989, biggest antenna farm I've ever seen!
>
>-Mike-

Yes Rugby is still on air, there has been a lot of new Rugby controlled clocks on sale
recently. I even saw a MSF Rugby gadget to connect to a LAN to synchronise all the
clocks on it. I have a Homebrew Rugby receiver but the QRM from the computers here
makes it next to useless unless I turn them all off and of course one of them decodes
the time signal! My receiver uses a ferrite rod (can't get much smaller than that!)

Regards Charles

By the way Rolf thanks for the URL about Turbo Codes, just need to work out how a
MAP decoder works now !

71. ARRL DCC and the old SS thread

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 8 Oct 96 12:37:05 +0100
To: hfsig@tapr.org
Message-ID: <H0000292028bea99@MHS>
Rendered body type: text/plain

> Warning: hot button alert!

> The proper measure of spectrum efficiency in a cellular environment is
> not QSOs per megahertz, but QSOs per megahertz PER SQUARE KILOMETER.
                                                  ****

> There's a very good reason (other than frequency stability and gain
> control reasons) why AMPS (analog) cellular uses FM instead of
> SSB. Thanks to its capture effect, FM can provide telephone quality
> audio with about 10-11 dB SNIR (signal-to-noise-plus-interference), as
> compared to about 25-30 dB for SSB.

I can't resist, sorry folks.

But I have to ARGEE with Phil :-)

SSB is more efficent for a single point to point link, at low audio qualities, but in many cases, especialy celular, this is not the case. Consider a celphone conversation where you could hear in the background someone elses conversation! For the wanted signal signal to be 30dB stronger, cells would need to be 10 time the diameter compared to FM which needs only 10dB, and thus encloses 100 times the number of users, so even if you can get 10 SSB users on a single FM channel, with FM you could reuse the chanel so that you could actually fit 100 FM users on the same channel in the same area, and with better audio quality, no cross talk, and cheaper phones.

Digital modulatoin has a similar capture effect, but uses less bandwidth than FM using standard compression (around 1/3), and half again with the new 'half rate' compression. No only that, but the phone uses less power, so you have longer talk times.

SS goes even further, allowing even more efficent modulation, even better reuse, and even dynamic compresion to given even more channel capacity.

I'd estimate SS would allow around 100 times the number of users in the same area with better audio quality and more features than SSB. Spectral efficency has got to be the *number one* reason for using SS in a cellular enviroment!

And these benifits CAN be made to work for amateur radio too! But they DON'T come for free.

Global HF radio could be compared to a huge single cell, and by breaking it up into smaller cells, reducing power and using interference tolerant systems such as SS, capacity could be dramatically increased, as in the example of the SSB celular system. Because of the vagueities of HF propagation the cellular structure would have to be dynamic but that's

within technological capability.

But the real question is not one of capacity, we have enough of that as it is, the real question is one of USEFULLNESS. It becomes a question of 'what is amateur radio about anyway'. Of course it's about experientation and learning, and even communication but that's more and more becomming a secondary function since modern communication networks are so good. So what use is the ham bands bands if you can only use a single mode, over local distances, and only do things that you could do much easier using the public networks? Maybe for the few who set up the system it would be great, but how many hams in a million is that? There would be so many aspects of ham radio that would be lost, that ham radio would simply cease to exist, since its only remaining use would be inferior to cheap public networks. Everyone who starts in radio must start somewhere, end even though the older modes are outdated and well behind the state of the art, they are simple and they are excelent for anyone how wants to learn about radio, basic principles never go out of date. You can learn a lot building and operating even a simple QRP CW rig, and there's nothing to stop you using modern DDS chips, micro controllers, power modules, RF IC's, etc etc. If I couldn't use RTTY (because of hopping impulses say) as a simple starting mode for my modem, BTL, I would have probably not even tried, now RTTY's going I can move on to more up-to-date modes.

SS might be more efficent than narrow band modes, but it is more efficent only when high power narrow band signals are excluded, and when SS uses the band in a way that is incompatible to any other form of amateur use of the bands, causing interference that it actually REDUCES THE USEFULLNESS of the amateur bands. IF SS must be interference resistant to get its capacity, then it MUST cause interference in acheiving its capacity, and dispite its resistance it is still not resistant enough to narrow band modes to acheive its capacity in their presence.

Certinly if interference tolerant, power controled modes became the norm, there would be many new applications that could use these modes, but there would be no point in putting them on HF radio, since HF radio would be reduced to the same coverage as conventional networks, but with much poorer performance.

I'm afarid I'm still as convinced as ever that SS cannot co-exist with narrow band modes, and the SS can never be the 'one true mode' in amateur radio, if amateur radio is to exist at all. I welcome SS as a new and fascinating facet of amateur radio, but am convinced that it must be used only in exclusive SS bands, especially the microwave bands where it is ideally suited to local, celular operation, and high, competitive, data rates.

For those interested, a NEW SS band on HF would be fun, but SS cannot presume to dictate to the rest of amateur radio.

Rob

Realisticaly PRO-SS!

Yes, I know there are SS sytems on HF & VHF already, but a few very wide band systems a long way from most HF users is quite different to a hundred SS users in every neighbourhood. Nothing is free, the band IS used, and if the theoretical capacity of SS is realised, then the bands will be just as heavilly loaded (apperaring to be more so to narrow band users since they use the extra capacity to get the quality and distance they expect). Even if it is only used as much as the bands are use right now, the loading on the band will still be significant to non-interference resistant modes.

72. Re: [HFSIG:1620] ARRL DCC and the old SS thread

From: RLANIER@mailb.harris.com (RLANIER)
Date: Tue, 8 Oct 1996 11:09:18 -0400
To: hfsig@tapr.org
Message-ID: <25a6ee50@mailb.harris.com>
Rendered body type: text/plain

This discussion of SS is getting pretty old. I hope to soon have a
complete DSSS system built so I can put some of these issues to rest.
SS has been used, is being used now, and will be used in future, even
more than it is today. WHY? Because it works effectively WITHOUT
causing harmful interference.

Keeping in mind that many companies are investing lots of money on

this subject. That alone should tell you something. If its not worth doing, then why are they doing it?????????????

There seems to be this pervasive paranoia about SS, almost as if its "invading" amateur radio and its going to take control of all the existing modes and strangle them out of existence!!! Man, some of you people are RIDICULOUSLY PARANOID!!

When the first high-speed SS network is built (and it will be built), then we shall see the amount of interference . But you probably won't know its there.

73s de
Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1620] ARRL DCC and the old SS thread
Author: hfsig@tapr.org at smtp
Date: 10/8/96 6:56 AM

> Warning: hot button alert!

> The proper measure of spectrum efficiency in a cellular environment is
> not QSOs per megahertz, but QSOs per megahertz PER SQUARE KILOMETER.
****

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> control reasons) why AMPS (analog) cellular uses FM instead of
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more becomming a secondary function since modern communication networks are so good. So what use is the ham bands bands if you can only use a single mode, over local distances, and only do things that you could do much easier using the public networks? Maybe for the few who set up the system it would be great, but how many hams in a million is that? There would be so many aspects of ham radio that would be lost, that ham radio

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73. [HFSIG:1621] Re: ARRL DCC and the old SS thread

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 8 Oct 96 18:52:27 +0100
To: hfsig@tapr.org
Message-ID: <H0000292028d5122@MHS>
Rendered body type: text/plain

> WHY? Because it [SS] works effectively WITHOUT causing harmful
> interference.

This statement needs to be qualified. It DOES work in MANY cases without harmfull interference, but the definition of 'harmfull' is highly dependent of the application and many other factors such as processing gain, propagation and encumbant applications.

Are there any successfull high speed SS systems that operate for example on the aeronaughtical VHF band near an airport with a range of several

miles, and several dozon similtainious users? My point is that existing applications have chosen their circumstances for a reason.

You don't get something for nothing, SS does not work by magic, it must still obey the laws of physics. It's a question of whether what you use will be missed.

Best of luck with your design, but there is potential for problems, and these are best overcome by forseeing them and designing the system around them. Its best not to design on blind faith. How will you overcome the problems of powerfull, nearby uncontrolled transmitters and nearby hidden receivers? Perhaps it just 'wont happen here'? Most systems rely on high processing gains, or short ranges, or channel hopping, or sole use of a band, or expect other users to be interference resistant. Whats the right mix for the band you will use? These choices will make or break a SS system, and if you intend widespread use, they are of even greater importance. Do you think designers of the various sucessfull systems ignored these considerations? Applying these considerations to the amateur HF situation gives a pretty grim picture, when the zealistic euphoria subsides.

Rob

74. Unsubscribe

From: Louis Dupree <cn1743@coastalnet.com>
Date: Tue, 08 Oct 1996 15:07:04 -0400
To: hfsig@tapr.org
Message-ID: <2.2.32.19961008190704.00683b4c@abaco.coastalnet.com>
Rendered body type: text/plain

How do I unsubscribe from this ? I tried and says I am not listed as having subscribed. I am going to be away for a while and not sure how much email my ISP wil hold for me. I can then re-subscribe when I get back. Thank you.
Louis

Louis J. Dupree Jr.
3015 Englewood Dr.
Kinston, NC 28504
ldupree@coastalnet.com

75. Re: [HFSIG:1622] Re: ARRL DCC and the old SS thread

From: RLANIER@mailb.harris.com (RLANIER)
Date: Tue, 8 Oct 1996 17:47:12 -0400
To: hfsig@tapr.org
Message-ID: <25acc2a0@mailb.harris.com>
Rendered body type: text/plain

I don't think SS works by magic - quite the opposite. What I get tired of hearing is this notion that SS will cause widespread interference. If run at 1500 watts, of course it will!!! But using automatic power control and proper design techniques, which most home brewers follow anyway, SS will not be this giant "killer" of the amateur bands.

Some hams are already using high-speed networks in various parts of the country and with good success. Why not use SS as well?? Of course there are obstacles to overcome - EVERY system has to do that. And hams CAN do it if we stop arguing about!!

I will continue to experiment with SS because I know it has a future. And if no one else wants to design a high-speed network, then I'll do it.

And then start my own company ...

73s de
Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1622] Re: ARRL DCC and the old SS thread
Author: hfsig@tapr.org at smtp
Date: 10/8/96 12:57 PM

> WHY? Because it [SS] works effectively WITHOUT causing harmful
> interference.

This statement needs to be qualified. It DOES work in MANY cases without harmfull interference, but the definition of 'harmfull' is highly dependent of the application and many other factors such as processing gain, propagation and encumbant applications.

Are there any successfull high speed SS systems that operate for example on the aeronaughtical VHF band near an airport with a range of several miles, and several dozon similtainious users? My point is that existing applications have chosen their circumstances for a reason.

You don't get something for nothing, SS does not work by magic, it must still obey the laws of physics. It's a question of whether what you use will be missed.

Best of luck with your design, but there is potential for problems, and these are best overcome by forseeing them and designing the system around them. Its best not to design on blind faith. How will you overcome the problems of powerfull, nearby uncontrolled transmitters and nearby hidden receivers? Perhaps it just 'wont happen here'? Most systems rely on high processing gains, or short ranges, or channel hopping, or sole use of a band, or expect other users to be interference resistant. Whats the right mix for the band you will use? These choices will make or break a SS system, and if you intend widespread use, they are of even greater importance. Do you think designers of the various sucessfull systems ignored these considerations? Applying these considerations to the amateur HF situation gives a pretty grim picture, when the zealistic euphoria subsides.

Rob

76. Re: [HFSIG:1620] ARRL DCC and the old SS thread

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 9 Oct 1996 02:18:25 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610090918.CAA27573@servo.qualcomm.com>
Rendered body type: text/plain

Since it seems the SS issue is coming up again, I'd like to address it in the larger context of where ham radio is going and where I think it has gotten off the track in the 25 years I've been licensed (I qualify for QCWA in November -- gotta sign up!)

>Global HF radio could be compared to a huge single cell, and by breaking

Depends on the band. Even 20m propagation is rarely worldwide -- skip zones and the like work to break it up. And propagation is decidedly regional on 40m and below, especially during the day. Reuse can and does take place on the HF bands. And it's in a reuse environment that SS really shines.

>are so good. So what use is the ham bands bands if you can only use a
>single mode, over local distances, and only do things that you could do
>much easier using the public networks? Maybe for the few who set up the

The same question could be asked for SSB and CW. What use is ham radio if not only have its public service aspects been rendered obsolete by advances in commercial communications, but the technology and amateur skill-set have also remained totally stagnant, 20 years and counting behind the commercial state of the art?

My basic philosophy in ham radio has always been to build open systems that can be taken apart, learned from, modified and improved by anyone who feels like it. I'm very disturbed by an almost complete conversion of the amateur market during the time I've been a ham to impenetrable, proprietary "black boxes".

In the old days one could expect the manual for any piece of gear you'd buy to include a complete schematic, as well as a comprehensive theory-of-operation section. Heathkits were the best. I used to spend hours poring over those schematics, learning whatever I could. I think I've always spent more time (and had more fun) doing that than actually operating the equipment!

I think the trend to the black box started when microprocessors first appeared in amateur gear. By itself this was a very good development, as writing and modifying software is an almost ideal ham activity; the capital costs are minimal compared to hardware, and the results are

easily shared.

But I don't know of a single major ham manufacturer that prints the microprocessor source code for their transceivers in the manual. They continued to provide schematics, but more and more of the functionality was now performed inside an impenetrable rectangle marked "CPU".

Even TAPR, an organization otherwise remarkable for its openness and commitment to amateur education, treated the source code to its TNCs as proprietary and gave it out only under license to manufacturers who paid substantial fees.

And now we have three major HF digital modes, Clover II, Pactor II and G-TOR. Not only are they all implemented solely in black boxes with unpublished source code, but even the modulation schemes themselves are largely unpublished. (At least the AX.25 protocol implemented by the proprietary TAPR TNC code was published).

Even worse, at least two of the three (Clover and G-TOR) are covered by granted or pending patents -- the absolute antithesis of the cooperative amateur spirit. (I've seen the Clover patent. As with most patents today, it describes only one tiny aspect of the real product -- the use of sequential carriers to combat ISI -- and gives almost no insight into the complete system. It apparently remains a trade secret.)

To be sure, this only mimics the usual practice in the commercial radio industry -- every Part 15.247 device I know of is a proprietary black box, heavily covered by patents. And many of them are crap (both the radios and the patents!)

I feel strongly that if ham radio is to come close to justifying its existence, we need to develop new modulation modes AND we must do it in a totally open, cooperative, GNU-like fashion. All waveform designs and source code made freely available, at least for amateur radio uses.

In other words, the goal of ham radio isn't just to develop and deploy the most advanced radio modulation techniques possible. That's already going on outside ham radio with far more talent and resources than we can possibly bring to bear. And you don't need a ham license to use them.

No, the real reason to develop and deploy new technologies in the amateur service is to make it possible for anyone interested to participate and perhaps learn something in the process.

No amateur would be forced to get his hands dirty with the details of spread spectrum, coding or whatever, and I am under no illusions that more than a small percentage of hams will want to. But it is vitally important to encourage that small percentage, because *they* will almost single-handedly justify ham radio's continued existence -- not the DXers, contesters or rag-chewers. Not that I have anything against those activites. But they simply won't pay the rent anymore.

So this means we have to do three things:

1. Set an example by designing and producing open, fully documented modulation schemes and the hardware and software to implement them, and encourage any interested parties to experiment, modify, contribute and learn;

2. Lobby for more rational licensing requirements that emphasize the increasing importance of relevant technical expertise in ham radio (i.e., abolishing the CW test for all license classes);

3. Lobby for the least restrictive rules possible for emission modes, bandwidths, etc, delegating all intra-service interference issues to the hams themselves instead of hard-coding them in the FCC rules.

I won't make any guarantees that no one will ever suffer interference from an experimental SS system. But who ever said we're entitled to any guarantees at all? I consider it a challenge to work these things out for ourselves. The FCC rules already let us do lots of things in theory that clearly could interfere with other amateurs, such as running 1.5KW of FM in the middle of the 2m satellite band. But by and large most hams are willing to cooperate, foregoing something that is technically legal in order to help another ham. (The few that don't

cooperate probably also ignore FCC rules anyway!)

Voluntary bandplans and gentlemen's agreements already exist on many
bands to promote a plurality of modes, and I'm confident that they can
be adapted to deal with spread spectrum. As can novel newer methods
for mitigating interference, such my idea of a standard narrowband
packet channel for dynamic local interference resolution. (If somebody
nearby is QRMing you on a particular frequency, you complain on the
packet channel and the interfering transmitter automatically QSYs,
QRPs or shuts up. Think of it as a general form of busy tone multiple
access.)

We're not directing air traffic, and we're not dispatching ambulance
and police services. We *are* supposed to be technical experimenters,
teachers and students. And if we aren't, we'll soon be nothing at all.

Phil

77. Re: [HFSIG:1625] Re: ARRL DCC and the old SS thread

From: Jim Sanford <jsanford@infi.net>
Date: Wed, 09 Oct 1996 07:05:54 -0400
To: hfsig@tapr.org
Message-ID: <325B8712.234B@infi.net>
Rendered body type: text/plain

Phil Karn wrote:
>

> But I don't know of a single major ham manufacturer that prints the
> microprocessor source code for their transceivers in the manual. They
> continued to provide schematics, but more and more of the
> functionality was now performed inside an impenetrable rectangle
> marked "CPU".
And some badly need improvement, like the documented problems with the
Yaesu 736R, and it's obsolescent, limited computer command set.

> And now we have three major HF digital modes, Clover II, Pactor II and
> G-TOR. Not only are they all implemented solely in black boxes with
> unpublished source code, but even the modulation schemes themselves
> are largely unpublished. (At least the AX.25 protocol implemented by the
> proprietary TAPR TNC code was published).
Which is why I refuse to subsidize their greed (and use of paying hams
to
beta test stuff they really want to sell to the military) by buying
their
stuff.
>
> Even worse, at least two of the three (Clover and G-TOR) are covered
> by granted or pending patents -- the absolute antithesis of the
> cooperative amateur spirit. (I've seen the Clover patent. As with
> most patents today, it describes only one tiny aspect of the real
> product -- the use of sequential carriers to combat ISI -- and gives
> almost no insight into the complete system. It apparently remains a
> trade secret.)
See recent issues of Dr. Dobb's Journal and C/C++ Users Group Magazines
for alarming and depressing news on the whole "software patents" issue.
Scary.

> I feel strongly that if ham radio is to come close to justifying its
> existence, we need to develop new modulation modes AND we must do it
> in a totally open, cooperative, GNU-like fashion. All waveform designs
> and source code made freely available, at least for amateur radio
> uses.
AMEN! Thank you and Johan for all that you're doing, openly.
>
> 1. Set an example by designing and producing open, fully documented
> modulation schemes and the hardware and software to implement them,
> and encourage any interested parties to experiment, modify, contribute
> and learn;
>
AMEN
Phil: As usual, well said.

73, Jim
wb4gcs@amsat.org

78. Re: [HFSIG:1623] Unsubscribe

From: Mike Murphree <n4cnw@ibm.net>
Date: Wed, 09 Oct 1996 07:37:22 -0400
To: hfsig@tapr.org
Message-ID: <325B8E72.45F3@ibm.net>
Rendered body type: text/plain

Louis Dupree wrote:
>
> How do I unsubscribe from this ? I tried and says I am not listed as having
> subscribed. I am going to be away for a while and not sure how much email
> my ISP wil hold for me. I can then re-subscribe when I get back. Thank you.
> Louis
>
> Louis J. Dupree Jr.
> 3015 Englewood Dr.
> Kinston, NC 28504
> ldupree@coastalnet.com

Just yesterday I saw a similar case where I was subscribed twice to
the same list under two different email addresses that would get
delivered to the same account, i.e. my previous provider uses both
of the following domains: pig.net and praxis.net. The fix in this
case was to find a mailer that explicitly defined the "From" field
in the headers and didn't let it default to what the provider picks
as default and unsubscribe with the proper address.

73
Mike N4CNW

79. Re: [HFSIG:1625] Re: ARRL DCC and the old SS thread

From: RLANIER@mailb.harris.com (RLANIER)
Date: Wed, 9 Oct 1996 08:46:34 -0400
To: hfsig@tapr.org
Message-ID: <25b9f7e0@mailb.harris.com>
Rendered body type: text/plain

The reason so many companies (and I assume TAPR as well) make software
     proprietary is to protect their product. You have to look at it from
     their point of view: a group or an individual has worked hard to
     produce a working product and they certainly DO NOT want another
     company or individual stealing that idea. This is a very real
     possibility because we live in a VERY competitive world. Giving out
     your secrets is like committing suicide.

     I can understand your frustrating, but believe me, if you had worked
     on a project only to see your competitor come out with a product with
     the same thing you had worked on, you would understand what I am
     talking about.

     Remember, companies are in the business to make money, not share their
     secrets with those who are curious!

     73s de
     Tony, KE4ATO

In the old days one could expect the manual for any piece of gear
you'd buy to include a complete schematic, as well as a comprehensive
theory-of-operation section. Heathkits were the best. I used to spend
hours poring over those schematics, learning whatever I could. I think
I've always spent more time (and had more fun) doing that than
actually operating the equipment!

I think the trend to the black box started when microprocessors first
appeared in amateur gear. By itself this was a very good development,
as writing and modifying software is an almost ideal ham activity; the
capital costs are minimal compared to hardware, and the results are
easily shared.

But I don't know of a single major ham manufacturer that prints the
microprocessor source code for their transceivers in the manual. They
continued to provide schematics, but more and more of the
functionality was now performed inside an impenetrable rectangle
marked "CPU".

Even TAPR, an organization otherwise remarkable for its openness and commitment to amateur education, treated the source code to its TNCs as proprietary and gave it out only under license to manufacturers who paid substantial fees.

And now we have three major HF digital modes, Clover II, Pactor II and G-TOR. Not only are they all implemented solely in black boxes with unpublished source code, but even the modulation schemes themselves are largely unpublished. (At least the AX.25 protocol implemented by the proprietary TAPR TNC code was published).

Even worse, at least two of the three (Clover and G-TOR) are covered by granted or pending patents -- the absolute antithesis of the cooperative amateur spirit. (I've seen the Clover patent. As with most patents today, it describes only one tiny aspect of the real product -- the use of sequential carriers to combat ISI -- and gives almost no insight into the complete system. It apparently remains a trade secret.)

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So this means we have to do three things:

1. Set an example by designing and producing open, fully documented modulation schemes and the hardware and software to implement them, and encourage any interested parties to experiment, modify, contribute and learn;

2. Lobby for more rational licensing requirements that emphasize the increasing importance of relevant technical expertise in ham radio (i.e., abolishing the CW test for all license classes);

3. Lobby for the least restrictive rules possible for emission modes, bandwidths, etc, delegating all intra-service interference issues to the hams themselves instead of hard-coding them in the FCC rules.

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channel and the interfering transmitter automatically QSYs, QRPs or
shuts up. Think of it as a general form of busy tone multiple access.)

We're not directing air traffic, and we're not dispatching ambulance
and police services. We *are* supposed to be technical experimenters,
teachers and students. And if we aren't, we'll soon be nothing at all.

Phil

80. Re: [HFSIG:1626] Re: ARRL DCC and the old SS thread

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Wed, 9 Oct 1996 08:20:14 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199610091320.IAA16110@www.kelly-afb.org>
Rendered body type: text/plain

In Jim's message he write:
>
> Phil Karn wrote:
[stuff deleted]
> Which is why I refuse to subsidize their greed (and use of paying hams
> to beta test stuff they really want to sell to the military) by buying
> their stuff.

Sorry Jim, I don't think they made their boxes for the Military
as the U.S., Nato and SE Asia militaries have better modems for
HF (and most are using them) than G-TOR, PACTOR-II or CLOVER II
have to offer. The MIL-STD (and Nato STD) is 4800/2400 BPS with
Reed-Soloman coding FEC, etc (MIL-STD 188-110c and later). There
are other MIL-STDs that also cover HF modems that are a bit different
than 188-110c.

> >
> > Even worse, at least two of the three (Clover and G-TOR) are covered
> > by granted or pending patents -- the absolute antithesis of the
> > cooperative amateur spirit.
[stuff deleted]
>
> > I feel strongly that if ham radio is to come close to justifying its
> > existence, we need to develop new modulation modes AND we must do it
> > in a totally open, cooperative, GNU-like fashion. All waveform designs
> > and source code made freely available, at least for amateur radio
> > uses.

AMEN and AMEN!

And I might add, we need to have these "run", if externally,
on boards like the TI and Motorola DSP boards and/or on
platforms like the TAPR DPS box. Or, to keep Phil happy,
run it all in software.

I think two thinks will happen if we do the above. One the
"experimentar/bulder" will take the "board" and make his own
computer/radio interface. Second, the OFs, like me, who can't see
to build anymore, will buy the board and interface from guys like
Gwyn Reedy. And everybody will be happy...even the League because
they can include a "real" construction project in the ARRL Manual.

> >
> AMEN! Thank you and Johan for all that you're doing, openly.
> >
> > 1. Set an example by designing and producing open, fully documented
> > modulation schemes and the hardware and software to implement them,
> > and encourage any interested parties to experiment, modify, contribute
> > and learn;
> >
> AMEN
> Phil: As usual, well said.
>
> 73, Jim
> wb4gcs@amsat.org
>

As that old black deacon from Waco, Texas used to say...
"Preach on brother, ya gotta an audiance".

73,

Walt/K5YFW

81. Why things are the way they are

From: Gwyn Reedy <gwyn@paccomm.com>
Date: Wed, 9 Oct 1996 12:57:46 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BBB5E1.791C7DC0@gwyn.paccomm.com>
Rendered body type: text/plain

Jim Sanford[SMTP:jsanford@infi.net] said, quoting Phil Karn:

>>Phil Karn wrote:
>>
>>
>> But I don't know of a single major ham manufacturer that prints the
>> microprocessor source code for their transceivers in the manual. They
>> continued to provide schematics, but more and more of the
>> functionality was now performed inside an impenetrable rectangle
>> marked "CPU".
>
>And some badly need improvement, like the documented problems with the
>Yaesu 736R, and it's obsolescent, limited computer command set.
>
>
>> And now we have three major HF digital modes, Clover II, Pactor II and
>> G-TOR. Not only are they all implemented solely in black boxes with
>> unpublished source code, but even the modulation schemes themselves
>> are largely unpublished. (At least the AX.25 protocol implemented by the
>> proprietary TAPR TNC code was published).
>
>Which is why I refuse to subsidize their greed (and use of paying hams to
>beta test stuff they really want to sell to the military) by buying their
stuff.
>
>
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>> by granted or pending patents -- the absolute antithesis of the
>> cooperative amateur spirit. (I've seen the Clover patent. As with
>> most patents today, it describes only one tiny aspect of the real
>> product -- the use of sequential carriers to combat ISI -- and gives
>> almost no insight into the complete system. It apparently remains a
>> trade secret.)
>
>See recent issues of Dr. Dobb's Journal and C/C++ Users Group Magazines
f>or alarming and depressing news on the whole "software patents" issue.
>Scary.
>
>
>> I feel strongly that if ham radio is to come close to justifying its
>> existence, we need to develop new modulation modes AND we must do it
>> in a totally open, cooperative, GNU-like fashion. All waveform designs
>> and source code made freely available, at least for amateur radio
>> uses.

>AMEN! Thank you and Johan for all that you're doing, openly.
>
>> 1. Set an example by designing and producing open, fully documented
>> modulation schemes and the hardware and software to implement them,
>> and encourage any interested parties to experiment, modify, contribute
>> and learn;
>
>AMEN
>Phil: As usual, well said.
>
>73, Jim
>wb4gcs@amsat.org

I feel the need to comment on Jim's greed comment. These comments are as
owner of one of the smaller manufacturing outfits and a long time ham
(Phil, I've only got 8 years till I qualify for Double QCWA).

First, I generally agree with most of Phil Karn's comments about the future
of ham radio and the attitudes and conditions which now exist that impede
progress. I do what I can to aid progress. Phil is extremely generous with
the results of his labors, but I would point out to Phil that he earns a
living at a large firm, and doesn't suggest that they should give their
product away.

While my company was not singled out as perpetrator of the situation

mentioned, we could have been. I would only hope that people would make judgements on an individual basis, and not indulge in group condemnation. And to do that, they need additional information. Licenses, confidentiality agreements, etc. limit what a company may do, even if they want to.

Secondly, I would hope that people could see beyond the (to me - false) perception that amateurs are somehow being victimized by companies which produce equipment for their use. Do you also feel victimized by your automobile or washing machine manufacturer? If not, then why by amateur equipment manufacturers? (If yes, then see your analyst.)

Perhaps my real bitch is what I perceive as the widespread lack of economic understanding in the amateur community and the American population in general. Having the costs of production and sales available to me certainly assures me that we are not operating in the realm of greed. Without publishing our entire financial statement, how does one convince someone else that they are not being ripped-off ?

If you live on a farm or grow a garden, you are probably upset about the food prices in grocery stores. You know how little you get paid for your product or how little it costs to grow (and you ignore the value of your time and labor). But people pay for convenience - local stores open 24 hours etc. The stores would go out of business if everyone made a garden, BUT THEY DON'T. And if they did, you would see much simpler food on the table.

Similarly, amateur equipment manufacturers exist because there is a market for what they produce at the price they charge. (And like farmers, they don't make much on it.) If there were no amateur manufacturers, there would be a lot more construction and experimenting, but much less of amateur radio in general. (Not sure that is bad, but that is another whole subject, as inappropriate for HFSIG as this message is, HI).

I don't mean to waste people's time reading this (nor to 'waste bandwidth' a wretched phrase) and certainly don't want to start a flame war. I just want to see more thoughtfulness and less shooting from the hip or conspiracy based comments.

Thank you for reading this.

Gwyn Reedy, W1BEL
PacComm Packet Radio Systems
http://www.paccomm.com

82. Re: [HFSIG:1630] Why things are the way they are

From: "Rick Williams" <rwilliam@cesa4.k12.wi.us>
Date: Wed, 9 Oct 1996 14:58:24 -0500
To: <hfsig@tapr.org>
Message-ID: <199610092000.PAA484744@gomer.wiscnet.net>
Rendered body type: text/plain

Gwyn Reedy had commented:

> Perhaps my real bitch is what I perceive as the widespread lack of economic
> understanding in the amateur community and the American population in
> general. Having the costs of production and sales available to me certainly
> assures me that we are not operating in the realm of greed. Without
> publishing our entire financial statement, how does one convince someone
> else that they are not being ripped-off ?
>
I do have seen this lack of understanding in the general population where folks dont have much of a grasp of the way the world works. But it really pains me when I see it in the amateur community who I like to think of as at least a little more knowledgeable.

Most comments about other peoples greed are a reflection of the very individual's own
greed (as it is with most negative comments about others). Those individuals who use this kind of language, rarely work for free themselves and often expect very nice salaries for the work they do.

But I heartily concur, that as a group, the amateur community should embrace new methods of communication and that is not happening. ARRL certainly has done little to promote this and the only organization that seems to be somewhat in that direction is TAPR.

My limited experience several years ago to try and get people excited about

new "high speed" digital was not appreciated very much. So there are
probably others like me who now sit on the sidelines and view what is going
on and hope for some direction from our ham organizations.

Rick, KV9U

83. Re: [HFSIG:1628] Re: ARRL DCC and the old SS thread

From: Greg Jones <wd5ivd@tapr.org>
Date: Wed, 9 Oct 1996 15:55:59 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199610092056.PAA16268@tapr.org>
Rendered body type: text/plain

>
> Even TAPR, an organization otherwise remarkable for its openness and
> commitment to amateur education, treated the source code to its TNCs
> as proprietary and gave it out only under license to manufacturers who
> paid substantial fees.

The reason for this is simple -- TAPR does not own the source for the TNC-2.
That is held by (now) InfoMotion which is kind enough to continue to provide
TAPR with the updates and allows us to make them available ot the amateur
community without anyroyalities going back to him.

The DSP-93 monitor code was held in the dark -- this was becuase we wanted
only one monitor for the system. One reason we have so many TNC-2 out in
the world is becuase the AX.25 code stayed stable...on the other hand we
still seem to be stuck at that speed and type system for the same types of
reasons.

Cheers - Greg, WD5IVD
Pres TAPR

84. Re: [HFSIG:1631] Re: Why things are the way they are

From: muphaus@cris.com (Marv Uphaus)
Date: Wed, 09 Oct 1996 20:13:41 -0500
To: hfsig@tapr.org
Message-ID: <F3EXyM82cm+Q085yn@cris.com>
Rendered body type: text/plain

On Wed, 9 Oct 1996 15:23:51 -0500 (CDT) "Rick Williams" <rwilliam@cesa4.k12.wi.us> wrote:

>Most comments about other peoples greed are a reflection of the very
>individual's own
>greed (as it is with most negative comments about others).

Woah, buddy...!!!

I cannot let your criticism of Phil go unanswered...!!!

First, you are generalizing here about whether someone of intelligence can
make a logical value judgement about another's motives without themselves
being guilty of the same sins... I think they can...!!! Otherwise the
publishing industry is out of business...!!!

I can also appreciate and understand Gwnn's comments about ham radio
manufacturers... The profit motive has to be there for good products to
continue to be developed... (A satisfied PacComm customer...)

Second, you obviously haven't been around digital ham radio for very long,
because if you had been, you'd realize that Phil Karn is one of the most
prolific and respected of the digital group... Phil has authored MANY
things (NOS being the most widely known) and given them to the ham radio
community... I'm sure Phil earns a nice salary with his company... But
his efforts for digital ham radio have been herculean over the years...

I quote Ian Wade, G3NRW, in his book, "NOSintro"...

"Without Phil's [free] contribution, it's unlikely that the amateur packet
network would be anything like as advanced as it is today."

As far as the development of digital modes, the marketplace will surely
shake out all those that do not work for amateur radio... Several are
dying as we communicate here on the Internet... I remember a few years
back, Phil commented that the Internet has taken over the bulk of digital
communications for the amateur community... He is certainly correct
there...!!! Otherwise we'd be doing this on 20 Meters...

And while I'm on my soapbox, let me say this...!!! If we want amateur

radio to flourish and grow, we will have to attract the technically
oriented kids of today... Ham radio is NOT doing that... All those young
people are heavily involved in computer projects, as we all were involved
in ham radio in our youth...

Keep up the good work, Phil...!!! All us old geezers (41 years licensed
now) appreciate your efforts and we also appreciate your comments... It's
good that you get us all rattled every once in a while...!!! ;-)

Marv, K4BVG...

Even when the experts all agree, they may well be mistaken.
                                  --Bertrand Russell
PGP PUBLIC KEY posted at pgp.mit.edu
---

85. SS et al..

From: LAITINEN@ESHOP.UOREGON.EDU
Date: Wed, 9 Oct 1996 22:36:03 -0700 (PDT)
To: hfsig@tapr.org, bcarver@magiclink.com
Message-ID: <961009223603.982@ESHOP.UOREGON.EDU>
Rendered body type: text/plain

Forwarded from Bill Carver, W7AAZ (ex-K6OLG)

Hey Larry....can you forward THIS as a comment?

After all is said and done, the ability of an amateur to interact with his
equipment on a hardware/software level is taken almost to the "impossible"
point in current equipment.

This total lack of visibility of the design makes it impossible for anyone
in our fraternity to improve any aspect of the situation. The lack of
visibility also appears to sometimes be used to "cover up" fairly serious
shortcomings and compromises that one would not expect from the glorious
advertising copy.

Lets cut through all the BS: TAPR can guide an open effort to develop and
optimize hardware/software for an efficient, narrowband HF data mode.
Lots of work has been done on point-to-point, high data rate
communications. I, for one, would like a mode suitable for
keyboard-keyboard, rountable, eavesdropping (ie, SWL) and autostart
functions.

Bill, W7AAZ

86. Re: [HFSIG:1622] Re: ARRL DCC and the old SS thread

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 10 Oct 1996 02:31:21 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610100931.CAA00427@servo.qualcomm.com>
Rendered body type: text/plain

>Are there any successfull high speed SS systems that operate for example
>on the aeronaughtical VHF band near an airport with a range of several
>miles, and several dozon similtainious users? My point is that existing
>applications have chosen their circumstances for a reason.

The military's JTIDS (Joint Tactical Information Distribution System)
is a spread spectrum system that operates in the 960-1215 MHz
TACAN/DME aeronautical navigation band on a non-interference basis. (I
know, that's UHF rather than VHF -- but it certainly *is* widely used
by civilian airliners for navigation.)

If I remember right, JTIDS notches out the 1030 and 1090 MHz slots in
its hopping sequence to protect secondary surveillance radars, but
they don't do anything special to protect the DMEs.

A quick Alta Vista search produced some Rockwell specs for some of
their JTIDS products. Data rates range from 28.8 to 238 kb/s.
Transmit powers up to 1000 W (!!) are available.

Perhaps Steve Bible can comment further on JTIDS.

--Phil

87. Re: [HFSIG:1628] Re: ARRL DCC and the old SS thread

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 10 Oct 1996 02:52:58 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610100952.CAA00445@servo.qualcomm.com>
Rendered body type: text/plain

The reason so many companies (and I assume TAPR as well) make software
     proprietary is to protect their product. You have to look at it from

That's what copyrights are for!

Believe it or not, I've done pretty well with commercial licensing of
NOS even though I put all of the source code out on the net. Enough to
help me pay off a good chunk of my mortgage. Sure, I've seen some
evidence of being ripped off, but there are enough honest companies
out there to make it worthwhile.

I admit, if NOS were my *sole* source of income I might feel somewhat
differently. In any event, I've been careful to avoid a very common
mistake: explicitly trying to make money off the ham market, as
opposed to doing it for fun and taking whatever comes later from
commercial spinoff sales as an unanticipated bonus.

I could name several who have made this mistake and bitterly regretted
it, but I don't think that's necessary. From the very beginning I knew
better than to even *think* of trying to charge hams for NOS or any
other software I do. As the saying goes, it'd be like trying to teach
a pig to sing. It wastes time and annoys the pig.

Phil

88. Re: [HFSIG:1633] Re: Why things are the way they are

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 10 Oct 1996 07:28:54 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199610101228.HAA20981@www.kelly-afb.org>
Rendered body type: text/plain

In Marv's message he writes:
>
> On Wed, 9 Oct 1996 15:23:51 -0500 (CDT) "Rick Williams" <rwilliam@cesa4.k12.wi.us> wrote:
>
> And while I'm on my soapbox, let me say this...!!! If we want amateur
> radio to flourish and grow, we will have to attract the technically
> oriented kids of today... Ham radio is NOT doing that... All those young
> people are heavily involved in computer projects, as we all were involved
> in ham radio in our youth...

        Boy is that the truth...just sign me as:

        Director for RF Communications,
        Young Astronaut Technology Program
        North East Independent School District (NEISD)
        San Antonio, Texas

        Roosevelt High School,
        TechMagnet High School for the NEISD

>
> Keep up the good work, Phil...!!! All us old geezers (41 years licensed
> now) appreciate your efforts and we also appreciate your comments... It's
> good that you get us all rattled every once in a while...!!! ;-)
>
> Marv, K4BVG...

        Marv, you're just a _young-buck_...ya gotta be over 55 to qualify
        as an OG or OF...
>
>

        73 all,

        Walt/K5YFW, ex KN5YFW

```
=====================================================================
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development.      |
=====================================================================
|                                  |                                  |
|                                  | The greatest dangers to liberty |
```

```
| Walt DuBose - K5YFW               | lurk in insidious encroachment  |
| E-Mail k5yfw@www.kelly-afb.org    | by men of zeal, well-meaning    |
| Business Telephone: (210)925-6081 | but without understanding.      |
|     Home Telephone: (210)696-3196 |                                 |
|                                   |   - Justice Louis D. Brandeis   |
|                                   |                                 |
=========================================================================

OBTW, if you've looked this far into my message, let me say that kids
like computer CW of HF more than voice...its an "HF chat room".

Please don't forget the JOTA, 19-20 Oct 96 on 80, 40?, 20, 15 & 10 meters.
See QST for frequencies but I think the 20 meter frequency is 14.290 MHz.
        -- k5yfw
```

89. Re: [HFSIG:1636] Re: ARRL DCC and the old SS thread

From: RLANIER@mailb.harris.com (RLANIER)
Date: Thu, 10 Oct 1996 08:49:28 -0400
To: hfsig@tapr.org
Message-ID: <25cf0be0@mailb.harris.com>
Rendered body type: text/plain

```
>>The reason so many companies (and I assume TAPR as well) make software
     >>proprietary is to protect their product. You have to look at it from

>That's what copyrights are for!

Why should a company file for a copyright to protect their product????
And copyrights expire!! Do you think Microsoft should file for a
copyright for DOS? Look, my point here is that a COMPANY has the right
to do what they want with their product. Just because we would like to
know how Kenwood or Yaesu used DSP techniques in their new receivers,
does not mean they have to show us how they did it!! Or to disclose
their source code.

Now, if an individual wants to spend his/her free time on a project and
then tell the rest of the amateur community about, great! But if they
decide, hey, I've worked very hard on this and I want to see something
in return, don't fault them for that either.

73s de
Tony
```

90. [HFSIG:1636] Re: ARRL DCC and the old SS thread

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 10 Oct 96 12:33:43 +0100
To: hfsig@tapr.org, karn@qualcomm.com
Message-ID: <H0000292028b3749@MHS>
Rendered body type: text/plain

```
>>The reason so many companies (and I assume TAPR as well) make software

>>proprietary is to protect their product. You have to look at it from
>
> That's what copyrights are for!

> Believe it or not, I've done pretty well with commercial licensing of
> NOS even though I put all of the source code out on the net. Enough to
> help me pay off a good chunk of my mortgage. Sure, I've seen some
> evidence of being ripped off, but there are enough honest companies
> out there to make it worthwhile.
>
> I admit, if NOS were my *sole* source of income I might feel somewhat
> differently. In any event, I've been careful to avoid a very common
> mistake: explicitly trying to make money off the ham market, as
> opposed to doing it for fun and taking whatever comes later from
> commercial spinoff sales as an unanticipated bonus.

Yip, thats my position on BTL as well. I have no costs to cover, and
write it out of interest, and to help stimulate amateur digital comms by
making it free and widely available. And hey, any fame and fortune that
comes my way is greatfully accepted!

I intent to release a description of the DSP side of BTL shortly (as
soon as I finish it!) and quite probably selected portions of the source
code that is of specific interest (not all the extra rubbish to do the
UI and all the other support rubbish). But focus on the soundblaster,
filtering, and synchronisation algorithems etc. Just the essential guts
of the source code. Right now the code is not very modular is a bit of a
```

mess, and that's holding me back in further developemnt of BTL, so when I get it sorted into modular units, I'll probably release specific modules so folks can see how its done.

It serves two purposes, firstly it makes the source code understandable, and usefull to anyone who wants to know how its done and perhaps wants to implement their own modem for whatever purpose or mode or platform etc, more for inspiration rather than drop-in directly useable code, and secondly, anything I may get from BTL is still protected, since its all the boring rubbish UI/support code that takes all the effort, and if anyone want to rewrite that, the result would be pretty much their own effort anyway, and anyone can do that. Users want a finished product.

Eluding back to the old SS thing..

I don't want to be seen as the spoil sport here, like most of you, I want to see experimentation in any new mode, but there are potential problems and they must be recognised and taken into account from the outset otherwise we will end up with a poor system that will not be very effective and may well cause interference to the point where the use of SS is so limited that its value is completely eroded. If we just build a DSSS system slap TCPIP over top, wait for the complaints and hope, we will quickly be locked into a system almost as hopeless as HF packet on 80m, except causing a lot more interference and agrovation.

SS is a mode with new properties and new rules, and the adhoc principles of sharing that allowed narrow band modes to coexist no longer work. The rules of the two systems are inherently different and what works for one does not nessesarily work for the other.

The absolutly worst thing we can do is blindly setup a system anywhere and just expect no interference out of pure faith.

We must DESIGN the system to work, and that means selecting appropriate bands, procesing gains, coding, antenna requirements, sensitivity requirements, repeater/node distribution, and data rates that WILL work. There were mistakes made with Packet, but the consequences weren't so bad. A bad SS system could not only spell then end of SS as a usefull mode, but also devalue ham radio to the point where perhaps it will cease to exist. The stakes are much higher, this issue must be taken seriously.

Once a bad system is running we will be locked into it an no amount of goodwill on behalf of the SS or narrowband users will be able to resolve the problems. By addressing these issues today we not only allay fears and gain support, but we will be getting the very best SS system we can get.

A while back I posted a set of guidelines intended as a first step along these lines on SS-SIG, but it appears gun-blazing faith and blindness is the prefered course of action. - This does nothing to instill confidance amounst the critics.

Rob

91. [HFSIG:1635] Re: ARRL DCC and the old SS thread

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 10 Oct 96 12:41:30 +0100
To: hfsig@tapr.org, karn@qualcomm.com
Message-ID: <H0000292028b3754@MHS>
Rendered body type: text/plain

> The military's JTIDS (Joint Tactical Information Distribution System)
> is a spread spectrum system that operates in the 960-1215 MHz
> TACAN/DME aeronautical navigation band on a non-interference basis. (I
> know, that's UHF rather than VHF -- but it certainly *is* widely used
> by civilian airliners for navigation.)
>
> If I remember right, JTIDS notches out the 1030 and 1090 MHz slots in
> its hopping sequence to protect secondary surveillance radars, but
> they don't do anything special to protect the DMEs.
>
> A quick Alta Vista search produced some Rockwell specs for some of
> their JTIDS products. Data rates range from 28.8 to 238 kb/s.
> Transmit powers up to 1000 W (!!) are available.

Now here's an example of a system thats obviously been well designed.

Navagation is fairly interference tollerant, 245 MHz -> 238kb/s is a lot
of processing gain, navagation ranges are a lot further than high speed
data links, the SS transmitters will be a long way from any airborne
navagation receiver, and of course they notch out sensitive bands.
Sounds like it should work, and it obviously does.

So lets apply so good design principles to the amateur application.

Rob

92. Higly robust HF modes

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 10 Oct 96 13:23:38 +0100
To: hfsig@tapr.org
Message-ID: <H0000292028b5f11@MHS>
Rendered body type: text/plain

> Lets cut through all the BS: TAPR can guide an open effort to develop
> and optimize hardware/software for an efficient, narrowband HF data
> mode. Lots of work has been done on point-to-point, high data rate
> communications. I, for one, would like a mode suitable for
> keyboard-keyboard, rountable, eavesdropping (ie, SWL) and autostart
> functions.
>
> Bill, W7AAZ

You are not alone :-)

There are a number of people interested in this application on HFSIG.

BPSK and MFSK have been suggested, I'd like to be able to spend more
time on it myself. Replacing RTTY with a more robust mode, but still
allowing the free form QSO's for contesting, roundtables etc is one
application, as is communicating with very poor signal to noise ratios,
such as QRP DX, emergancy comms from a handheld HF rig and simple low

wire antenna, and the LF band (73kHz in the UK).

I wonder if there has been any recent progress on the BPSK front?

Cheers,

Rob

93. Re: [HFSIG:1639] Re: ARRL DCC and the old SS thread

From: RLANIER@mailb.harris.com (RLANIER)
Date: Thu, 10 Oct 1996 14:31:17 -0400
To: hfsig@tapr.org
Message-ID: <25d40e50@mailb.harris.com>
Rendered body type: text/plain

Eluding back to the old SS thing..

>SS is a mode with new properties and new rules, and the adhoc principles
>of sharing that allowed narrow band modes to coexist no longer work. The
>rules of the two systems are inherently different and what works for one
>does not nessesarily work for the other.

>The absolutly worst thing we can do is blindly setup a system anywhere
>and just expect no interference out of pure faith.

>We must DESIGN the system to work, and that means selecting appropriate
>bands, procesing gains, coding, antenna requirements, sensitivity
>requirements, repeater/node distribution, and data rates that WILL
>work. There were mistakes made with Packet, but the consequences
>weren't so bad. A bad SS system could not only spell then end of SS as
>a usefull mode, but also devalue ham radio to the point where perhaps
>it will cease to exist. The stakes are much higher, this issue must be
>taken seriously.

> You're absolutely right, Rob. SS definitely has to be carefully
> thought out in order to work correctly. I don't think one single bad
> design will cause amateur radio to cease to exist. It would definitely
> give the non-SS crowd some fuel for their fire, though.
>
> Some have argued that using 220MHz and above is the way to go. The
> only benefit I can see for this is having a wider spreading band. One
> important issue is the use of error correction, which will become very

important in high-speed data transmission. Another is data conversion, but this is becoming a moot issue with the new ADCs and DACs coming out. I for one will keep slugging away at it and yes, I will share what I come up with!

>A while back I posted a set of guidelines intended as a first step along
>these lines on SS-SIG, but it appears gun-blazing faith and blindness is
>the prefered course of action. - This does nothing to instill confidance
>amounst the critics.

>Rob

I think the reason for this is there is still widespread discussion about the merits of SS in general - interference being the biggest concern. Some of us will have to start discussing certain aspects of SS and go from there.

73s de
Tony, KE4ATO

94. Re: [HFSIG:1641] Higly robust HF modes

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 10 Oct 1996 14:37:19 -0800
To: hfsig@tapr.org
Message-ID: <199610102139.OAA21906@PEAK.ORG>
Rendered body type: text/plain

Hi Rob,

Congratulations on BTL! I have played with it and you have done a nice job. Besides the fact that it works better than a lot of TNC's on RTTY - you seem to have mastered the art of dealing with all those Sound-Blaster compatibility issues.

There recently was another announcement by Brian Cauchi, 9H1JS, of his program called FTV. It incorporates SSTV, WEFAX, and RTTY - all with a Soundblaster card. Looks like a nice job, but doesn't run on my Soundblaster which is a genuine Creative Labs, SB16. So, that was a bit of a dissapointment. See http://www.geocities.com/SiliconValley/2504/ for a link to the FTV software.

>BPSK and MFSK have been suggested, I'd like to be able to spend more
>time on it myself. Replacing RTTY with a more robust mode, but still
>allowing the free form QSO's for contesting, roundtables etc is one
>application, as is communicating with very poor signal to noise ratios,
>such as QRP DX, emergancy comms from a handheld HF rig and simple low
>wire antenna, and the LF band (73kHz in the UK).
>
>I wonder if there has been any recent progress on the BPSK front?
>

Yes, as a matter of fact. There is a small group for G and PA0 hams that now uses a slowspeed BPSK (31.5 baud). Peter Martinez, G3PLX, devised a novel coding scheme for this keyboard-keyboard mode that may well replace regular RTTY as we know it. It also has the possibility to be implemented using the Soundblaster approach as for BTL. Here is how its works:

"The idea is very like morse-code, which has the interesting property that
although it's an asynchronous code, it can never stay out of character-sync
on single bit-errors, since the 000 letterspace pattern never occurs within a codeword. Once I spotted that, the variable-length code almost devised itself:

1) Codewords are separated from each other by two consecutive zeros.
2) No codeword contains 2 or more embedded consecutive zeros.

It follows from this that all codewords start and end with 1.

I collected about 1M of assorted (english) ASCII(128) text files and sorted them into descending frequency order, then assigned them to a list
of all binary codewords meeting the above two criteria, in ascending order
of length. The result is the alphabet at the end of this message, which is
in ASCII order. The longest code is 10 bits. I transmit LSB first. The

code shown in the alphabet has the 00 gap added on the end (the msb end) and
then a dummy '1' which marks the length and is not transmitted. The codeword
for the letter 'a' is, for example, 1101, transmitted left to right.
idle-line is a continuous 00000 stream, and of course with BPSK I choose this to be represented by continuous phase-reversals so that the distant receiver can recover bitclock. There are never more consecutive 1's than the maximum-length codeword, so bitstuffing isn't needed."

(Personal Communication: Peter Martinez, G3PLX)

Neat - is'nt it? I'll ask for Peter's consent to post the code alphabet and a pseudo coder/decoder and get more folks involved.

--Johan

95. Re: [HFSIG:1622] Re: ARRL DCC and the old SS thread

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 10 Oct 1996 17:04:51 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610110004.RAA01942@servo.qualcomm.com>
Rendered body type: text/plain

>You don't get something for nothing, SS does not work by magic, it must
>still obey the laws of physics. It's a question of whether what you use
>will be missed.

I actually quite agree with this. SS systems have to be engineered just like any other communication system. Some will work and some won't, and some will interfere just as some won't. It all depends.

Where I think we part company is in what the rules should say. I advocate the least restrictive FCC rules possible on SS, with the amateur communitity working out its own technical standards and procedures to resolve intra-service QRM. It is simply not realistic or fair to maintain a flat ban on SS on so many amateur bands because of the possibility (not certainty) of QRM. Again, we're not a "safety of life" service, we're an experimental and educational service. We need rules that recognize this, or we're going to fade completely into irrelevance.

By the way, you sound much like Andrew Viterbi in an informal paper he wrote for IEEE Communications Magazine back in April 1985. Apparently he too was perturbed at the time by the overzealousness of some SS proponents. But he probably regretted ever writing that paper when his own words were quoted back to him in the early days of Qualcomm CDMA by those who thought that the relative merits of CDMA could be debated by analogy and appeals to authority instead of being resolved by quantitative analyses and actual experiments.

Phil

96. Re: [HFSIG:1644] Re: ARRL DCC and the old SS thread

From: RLANIER@mailb.harris.com (RLANIER)
Date: Fri, 11 Oct 1996 08:49:54 -0400
To: hfsig@tapr.org
Message-ID: <25e42300@mailb.harris.com>
Rendered body type: text/plain

>Where I think we part company is in what the rules should say. I
>advocate the least restrictive FCC rules possible on SS, with the
>amateur communitity working out its own technical standards and
>procedures to resolve intra-service QRM. It is simply not realistic or
>fair to maintain a flat ban on SS on so many amateur bands because of
>the possibility (not certainty) of QRM. Again, we're not a "safety of
>life" service, we're an experimental and educational service. We need
>rules that recognize this, or we're going to fade completely into
>irrelevance.

>Phil

I couldn't agree more! Whats really worrysome is the blind rage some amateurs have when it comes to SS. I can parallel it to the current state of our federal government, but that is another subject ...

I'm not going to argue the merits of SS anymore. I KNOW its a good modulation method. So I will proceed with the ideas I have. And yes, I will share them with the rest of the amateur community ;)

73s de
Tony, KE4ATO

97. Re: ARRL DCC and the old SS thread

From: "Breon, Roy K" <rkb1@rfpo1.rfc.comm.harris.com>
Date: Fri, 11 Oct 96 09:09:00 DST
To: "'smtp:hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <325E45D8@smtpgate.rfc.comm.harris.com>
Rendered body type: text/plain

There have been several comments on this list recently regarding proprietary waveforms and/or proprietary software/firmware. I just thought that I'd add my $.02 from a manufacturer's perspective.

Harris Corporation is the inventor of the high speed HF modem technology that was later implemented into MIL-STD-188-110 and some of its appendices, As Walt DuBose stated in an earlier post, these specifications relate to speeds up to 4800 BPS in a 3kHz HF channel. The specifications define the on -the-air waveform characteristics such that many people could generate and receive the waveform. We released that information on the waveform design and do not retain any patent rights in this area. This was done in the interest of promoting advanced used of HF radio and to encourage others to use such modems. Since then several manufacturers introduced MIL-STD-188-110 modems with various degrees of compliance to the specification. The techniques and source code that we use to generate the waveform is not covered in the specification and is not released in any of our instruction manuals either. It is copyrighted and also retained as a trade secret. In fact, the details have changed over the years as newer hardware technology became available. The earliest versions of the RF-5254 HF Modem used bit slice technology. Newer versions such as the RF-5710 HF Modem use DSP technology.

The receiver portion, the DEM portion of the MODEM, is where the real expertise lies. The MIL spec does not really dwell on the demodulation portion. You might expect this since the spec is only concerned with interoperability and that basically involves the on-the-air waveform. We do have several patents related to how the receiver portion of our modems is implemented. Harris provides some pretty good features such as filtering out jamming (read interfering) waveforms, adaptive channel equalization, data directed equalization, and soft decision decoding, etc. These features are not described in the MIL spec but are used to wring out as much information as possible from the standardized transmitted waveform. However, others could demodulate the waveforms using other techniques without infringing on our patents. The demodulator just wouldn't be as good and wouldn't perform with as good a BER for a given Eb/No and other channel conditions. In addition to the patent and copyright protection, the exact source code is retained as proprietary information.

Speaking of interfering signals -- noise is a form of interference. Using our modem, we can actually receive error free data with received signals having negative signal to noise ratios. In other words, the signal is below the noise. This has obvious benefits for the user.

One could consider some of these advanced waveforms as a form of spread spectrum. The serial tone waveforms described by the Mil spec use a single tone modulated in a very high and complex way. For slow speed waveforms such as 75 bps which could be transmitted in a fairly narrow bandwidth, we continue to use the full 3 kHz channel thus "spreading" the waveform.

Development of such technology is not inexpensive nor is the cost to produce such a product. In order for Harris and other companies to protect their investment it is necessary to keep much of the implementation technology proprietary. As you all know, unlike duplicating hardware, firmware that costs hundreds of thousands of dollars to create can be easily duplicated for a very low cost. Therefore, if development in the field is to continue, it is mandatory that patents, copyrights, and trade secrets be used.

Roy K. Breon
Manager, Marketing
Harris Corporation
RF Communications Division

98. Re: [HFSIG:1646] Re: ARRL DCC and the old SS thread

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 11 Oct 1996 10:15:28 -0800
To: hfsig@tapr.org
Message-ID: <199610111717.KAA17821@PEAK.ORG>

Rendered body type: text/plain

Roy,

I really enjoyed and appreciated your posting regarding Harris' products.

..... (lines deleted)....

>Development of such technology is not inexpensive nor is the cost to produce
>such a product. In order for Harris and other companies to protect their
>investment it is necessary to keep much of the implementation technology
> proprietary. As you all know, unlike duplicating hardware, firmware that
>costs hundreds of thousands of dollars to create can be easily duplicated
>for a very low cost. Therefore, if development in the field is to continue,
>it is mandatory that patents, copyrights, and trade secrets be used.
>
>Roy K. Breon
>Manager, Marketing
>Harris Corporation
>RF Communications Division
>

I have no problem with this philosophy. There are those ways and means for maufacturers to protect their technolgy. Regarding amateur radio experimentation on the other hand; I have personally tried to make the point at all HFSIG meetings that we are not asking, nor expecting proprietory trade secrets or software from anyone (it would be nice, but naive to expect that :-).

The troubling issue, in my mind, is (was) the recent emergence of proprietory protocols that excludes all amateurs from experimenting with basic principles. This is contrary to the spirit of the amateur radio hobby. If one is into manufacturing or into some commercial venture, you should expect to follow the rules, pay the price and reap the rewards, thats another matter - commercial radio. Fortunately, Part 97 now outlaws such unpublished protocols for use on the amateur radio bands. So we may see things change for the future.

We as amateur experimentors, can, and must continue to participate in the development of future HF digital communication systems. I believe that this kind of grassroots effort is possible from the talents and synergy from a group like HFSIG. It will, however, only be a small handful that will actually do the effort to translate casual discussion into real working models. I am not sure that these folks pose much, if any, threat to manufacturers and business.

My 2-cents worth.

--Johan

99. Re: [HFSIG:1646] Re: ARRL DCC and the old SS thread

From: Charles Brain <chbrain@dircon.co.uk>
Date: Fri, 11 Oct 1996 17:50:15 +0100
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19961011165015.00678a00@popmail.dircon.co.uk>
Rendered body type: text/plain

>have several patents related to how the receiver portion of our modems is
>implemented. Harris provides some pretty good features such as filtering
>out jamming (read interfering) waveforms, adaptive channel equalization,
>data directed equalization, and soft decision decoding, etc. These features

Yes as Fredricks found out to their cost!

>
>Roy K. Breon
>Manager, Marketing
>Harris Corporation
>RF Communications Division
>
>
>

Charles Brain.

100. Re: [HFSIG:1630] Why things are the way they are

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Mon, 14 Oct 1996 02:32:56 -0700 (PDT)

To: gwyn@paccomm.com
Cc: hfsig@tapr.org
Message-ID: <199610140932.CAA14733@unix.ka9q.ampr.org>
Rendered body type: text/plain

Gwyn,

Some comments on your reply to my message of last week.

I certainly don't take you, Paccomm or any other manufacturer to task for charging for the hardware products you sell. Companies like Paccomm provide a necessary service to the amateur community. And since you take a financial risk whenever you bring a product to market, it's entirely reasonable to make a profit in exchange for that risk. My problem is most definitely NOT with the fact that commercial amateur products cost money.

My problem *is* that they come so often as black boxes. Sure, they generally work as advertised. But they generally fail to perform a more important function, which is to give the amateur who buys them the opportunity to learn as much as he wants about how the device functions.

There may or may not be a schematic. But there is rarely a source code listing, and with so much of the functionality of today's products in software a schematic doesn't say very much.

I understand that the TAPR TNC code was not yours (or even TAPR's) to publish. That was certainly the perogative of the original author. I am only pointing out the drawbacks of that position as seen from the larger perspective of what the amateur service is supposed to be all about -- self-motivated, hands-on, technical education and training.

I guess it is just a sign of the times that most hams don't seem to care what's behind the knobs (or "underneath the keyboard", to be more current.)

But I do care very much. The non-ham world has resources to develop new communications technology that we couldn't possibly match (even if we weren't already 20 years behind), cell phones and other commercial systems have made ham radio all but irrelevant for emergency communications, and the Internet has made ham radio essentially irrelevant in the "international good will" category.

So I see very clearly the fact that technical education is the only aspect of ham radio that still has any relevance, and black boxes don't exactly promote this particular goal.

Phil

101. SS thread

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 15 Oct 96 18:45:15 +0100
To: hfsig@tapr.org
Message-ID: <H0000292029241b2@MHS>
Rendered body type: text/plain

> You're absolutely right, Rob. SS definitely has to be carefully
> thought out in order to work correctly. I don't think one single bad
> design will cause amateur radio to cease to exist. It would definitely

> give the non-SS crowd some fuel for their fire, though.
>
> Some have argued that using 220MHz and above is the way to go. The
> only benefit I can see for this is having a wider spreading band. One
> important issue is the use of error correction, which will become very

> important in high-speed data transmission. Another is data conversion,

> but this is becoming a moot issue with the new ADCs and DACs coming
> out. I for one will keep slugging away at it and yes, I will share
> what I come up with!

> I think the reason for this is there is still widespread discussion
> about the merits of SS in general - interference being the biggest
> concern. Some of us will have to start discussing certain aspects of
> SS and go from there.

I agree. But I believe the two are strongly related. There is no doubt a poor system can cause interference. Many see the claimed advantages

weighed against this interfernce potential and don't like the way the balance swings, the see too few benifits for too few people and a great risk to them personally. The only way to convince people that experimenting in SS is worthwhile (which I believe it is) is to reassure them that there will be no interference. And I dont mean 'because I say so' or 'other systems don't have problems', that doesn't wash. What we need to a set of guidelines that outline the bare bones of a system (or systems) that on paper will not cause interference.

Sure, the practicalities may work out in our favor, but thats no gaurantee. We must start off small, and err on the side of caution to convince people we are serious about non-interference. Publishing guidlines and arguments why there will be no interference if these guidelines are followed, as well as solutions or even work being done to find solutions to the problems will go some way to convincing people that we are not out to take away their enjoyment of the bands, and do take non-interference seriously.

When we understand more about what makes a system work in an amateur enviroment and have solved the various problems it presents then we can perhaps move on to other bands, greater data rates, and more intensive sharing.

I believe a lot can be acheived with quite restricted systems that non SS people can feel comfortable with.

Paricularly at UHF/microwave frequencies which are ideally suited to low interference. These frequencies have many advantages:

- High processing gains and thus low PSD's and high tollerance to uncontrolled transmitters

- Directional antennas which in turn allows lower power, and even less in unintended directions

- Low band occupancy, ie the chances are that your neighbour won't be using the band. - unless they're using SS to, where normal near-far control will do the trick. (central coordination, band splitting (TX/RX), power control)

- Interference tollerant encumbant modes, ie FM

- Encumbant users typically use directional antennas further reducing interference potential

- Other users also use low power, reduceing interference to a SS system

- Propagation is natuarly limited, reducing interfernce potential and the number of possible SS/narrow band stations that could cause interference to either system.

- They're politically easier, since it wont effect the bulk of the amateur population. It's easier to coordinate interference reduction efforts with a few local users than with many users on a national or global scale.

- Competitive, high data rates are possible for the same processing gain, although more gain will probably be needed to counter the effects of increasing power to acheive the same Eb/No. Still an overall gain.

These are the ideas that need to be thrashed out. General discussion serves only to raise awareness of problems unless the issue of interference is effectivly dealt with at the same time.

Rob

102. [HFSIG:1643] Re: Higly robust HF modes

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 15 Oct 96 19:04:44 +0100
To: forrerj@peak.org, hfsig@tapr.org
Message-ID: <H0000292029241bd@MHS>
Rendered body type: text/plain

> See http://www.geocities.com/SiliconValley/2504/ for a link to the FTV

> software.

I've downloaded it. I'll give it a try.

G3PLX wrote:

> "The idea is very like morse-code, which has the interesting property
> that although it's an asynchronous code, it can never stay out of
> character-sync on single bit-errors, since the 000 letterspace pattern
> never occurs within a codeword. Once I spotted that, the variable-
> length code almost devised itself:

I like it. Does it also have the property that it works just as well if
the polarity is reversed, ie you can detect that polarity is reversed,
flip it and carry on decoding OK. Or does it require a differential BPSK
channel, ie change/no-change where polatity is always known?

> Neat - is'nt it? I'll ask for Peter's consent to post the code
> alphabet and a pseudo coder/decoder and get more folks involved.

Yes please.

Thanks,

Rob

103. Ham Radio Clip Art

From: Larry Bush <lbush@eramp.net>
Date: Tue, 15 Oct 1996 14:51:09 -0500
To: hfsig@tapr.org
Message-ID: <2.2.32.19961015195109.006f93b0@eramp.net>
Rendered body type: text/plain

Does anyone know of a source of Ham Radio Clip Art?
I Want to make up my own QSL and calling Cards.

73

Larry Bush, W5NCD
359 Arrowhead Point
Waco, TX 76712
Ph. 817-848-5155

104. Re: [HFSIG:1650] SS thread

From: RLANIER@mailb.harris.com (RLANIER)
Date: Tue, 15 Oct 1996 15:55:13 -0400
To: hfsig@tapr.org
Message-ID: <263eccd0@mailb.harris.com>
Rendered body type: text/plain

>so' or 'other systems don't have problems', that doesn't wash. What we
>need to a set of guidelines that outline the bare bones of a system (or
>systems) that on paper will not cause interference.

     Why not follow the existing amateur rules that all ready exist? We
     shouldn't cause harmfull interference, period. Neither should someone
     running a SSB system at 100 watts. If we follow good design practices
     and workmanship, which all homebrewers should, then we shouldn't have
     any problems.

>Publishing guidlines and arguments why there will be no interference if
>these guidelines are followed, as well as solutions or even work being
>done to find solutions to the problems will go some way to convincing
>people that we are not out to take away their enjoyment of the bands,
>and do take non-interference seriously.

     I think what would work even better is if a club or group would
     conduct tests of SS and "regular" systems, side-by-side, and listen
     for interference. What better way than to demonstrate it?

>When we understand more about what makes a system work in an amateur
>enviroment and have solved the various problems it presents then we can
>perhaps move on to other bands, greater data rates, and more intensive
>sharing.

     Of course. You always want to start small, test your idea and go from
     there. Its a "bottom-up" design approach.

     My approach is to use a chip set, design the system around that, use
     low output power (I believe the PRISM chip set outputs 250 mW) and go
     from there. Once that works, design a more "fancier" system, using
     frequency hopping, hybrid SS and faster data rates.

73s de
Tony

105. Re: [HFSIG:1638] Re: ARRL DCC and the old SS thread

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 15 Oct 1996 15:12:37 -0700 (PDT)
To: RLANIER@mailb.harris.com (RLANIER)
Cc: hfsig@tapr.org
Message-ID: <199610152212.PAA12455@servo.qualcomm.com>
Rendered body type: text/plain

>Why should a company file for a copyright to protect their product????
>And copyrights expire!! Do you think Microsoft should file for a
>copyright for DOS? Look, my point here is that a COMPANY has the right

Every copy of DOS I've ever seen has a Microsoft copyright notice. And
current US law says that copyrights are automatic; you don't have to
file for one unless you want to sue for infringement.

Copyrights have a term of the life of the author plus 50 years,
or for "works for hire", 100 years from the date of creation or 75
years from first publication, whichever comes first.

These terms are so long, especially for computer software, as to be
essentially infinite. (Will a copy of DOS have any real value in 75
years?) Such lengthy terms mock the Constitutional phrase "for a
limited time" as used in the section that authorizes patents and
copyrights. Just recently Congress considered extending the term to an
even more absurd 100 years.

Perhaps you're thinking of patents? Until recently, patent terms were
17 years from grant. Now they're 20 years from filing or 17 years from
grant, whichever is longer. Although these terms are shorter than
copyright terms, given the rapid pace of modern technology they are
still far too long.

Making patents even more of a problem is the abysmal state of the
patent system. Its examiners (most of whom are substandard engineers
unable to get jobs in real life, and/or lawyer wannabees getting a
free law school education from Uncle Sam) routinely grant thousands of
patents on obvious, trivial "inventions" or on things for which
abundant prior art exists. Companies routinely exploit this sad state
of affairs to create "landmines" to sabotage legitimate competition,
as opposed to protecting any substantial investments of their own.

The most important (and by far the most odious) characteristic of a
patent is that independent reinvention is not a defense to a claim of
infringement; you can be denied the right to use your own completely
independent work. The laws of physics and mathematics are the same for
everyone, and throughout history they (and the inventions that follow
from them) have often been independently (re)discovered by many people
at around the same time. For the law to grant exclusive "ownership" of
laws of nature to those with the best paid lawyers is the height of
human arrogance. (Yes, I know that in theory, only inventions, not
physical laws can be patented, and that patent rights are supposed to
go to the first inventor, not to the one with the most money. But
the practice is vastly different from the theory.)

I didn't really want to get into a deep discussion of intellectual

property on this list, but as you can probably tell by now this is one
of my hot buttons. For more on the trouble with patents, see my web page

http://www.qualcomm.com/people/pkarn/patents.html

--Phil

106. Re: [HFSIG:1636] Re: ARRL DCC and the old SS thread

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 15 Oct 1996 15:15:49 -0700 (PDT)
To: Robert.Glassey@nmp.nokia.com
Cc: hfsig@tapr.org
Message-ID: <199610152215.PAA12458@servo.qualcomm.com>
Rendered body type: text/plain

>A while back I posted a set of guidelines intended as a first step along
>these lines on SS-SIG, but it appears gun-blazing faith and blindness is
>the prefered course of action. - This does nothing to instill confidance
>amounst the critics.

By chance are you going to the AMSAT annual meeting in Tucson? I
signed up to give a talk on spread spectrum, and I'm thinking I ought
to orient it to a discussion of the points you've been making
recently. I could title it something like "When (and When Not) To
Spread", along the lines of the Viterbi papers in IEEE Communications
magazine over the past 15 years or so.

Phil

107. Re: [HFSIG:1646] Re: ARRL DCC and the old SS thread

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 15 Oct 1996 17:00:02 -0700 (PDT)
To: "Breon, Roy K" <rkb1@rfpo1.rfc.comm.harris.com>
Cc: hfsig@tapr.org
Message-ID: <199610160000.RAA12677@servo.qualcomm.com>
Rendered body type: text/plain

>Speaking of interfering signals -- noise is a form of interference. Using
>our modem, we can actually receive error free data with received signals
>having negative signal to noise ratios. In other words, the signal is below
>the noise. This has obvious benefits for the user.

This statement is either wrong or meaningless. What does "negative
signal to noise ratio" mean? If you mean Eb/N0 ratios below -1.6 dB, I
have this bridge in New York you might be interested in buying. Rumor
has it that it's going to be named after Claude Shannon.. :-)

But if you mean a negative signal-to-noise ratio in the operating
bandwidth, then your statement is meaningless without specifying both
the operating bandwidth and the user data rate. It's easy to make a
modem that operates at "negative signal-to-noise ratios" when the
bandwidth is much higher than the data rate -- spread spectrum and low
rate FEC are two well-known ways to do this.

That's why it's much better to just drop the notion of "signal to
noise ratio" when discussing modem performance and talk strictly about
Eb/N0 ratios, because they quantify the performance of a demodulator
in a way that's independent of bandwidth and data rate.

You can compute Eb/N0 from SNR, bandwidth and data rate with the following
formulas:

Eb/N0 = (S/R)/(N/B) = (S/N)/(R/B)

Eb = energy per bit, joules
N0 = noise spectral density, watts/hz (joules)
S = signal power, watts
R = user data (*NOT* coded symbol) rate, bits/sec
N = noise power, watts
B = signal bandwidth, hz

--Phil

108. Re: [HFSIG:1654] Re: ARRL DCC and the old SS thread

From: Jim Sanford <jsanford@infi.net>
Date: Tue, 15 Oct 1996 20:46:52 -0400
To: hfsig@tapr.org
Message-ID: <3264307C.433@infi.net>
Rendered body type: text/plain

Phil Karn wrote:
>
> >Why should a company file for a copyright to protect their product????
> >And copyrights expire!! Do you think Microsoft should file for a
> >copyright for DOS? Look, my point here is that a COMPANY has the right
>
> Every copy of DOS I've ever seen has a Microsoft copyright notice. And
> current US law says that copyrights are automatic; you don't have to
> file for one unless you want to sue for infringement.
>
> Copyrights have a term of the life of the author plus 50 years,
> or for "works for hire", 100 years from the date of creation or 75
> years from first publication, whichever comes first.

>
> These terms are so long, especially for computer software, as to be
> essentially infinite. (Will a copy of DOS have any real value in 75
> years?) Such lengthy terms mock the Constitutional phrase "for a
> limited time" as used in the section that authorizes patents and
> copyrights. Just recently Congress considered extending the term to an
> even more absurd 100 years.
>
> Perhaps you're thinking of patents? Until recently, patent terms were
> 17 years from grant. Now they're 20 years from filing or 17 years from
> grant, whichever is longer. Although these terms are shorter than
> copyright terms, given the rapid pace of modern technology they are
> still far too long.
>
> Making patents even more of a problem is the abysmal state of the
> patent system. Its examiners (most of whom are substandard engineers
> unable to get jobs in real life, and/or lawyer wannabees getting a
> free law school education from Uncle Sam) routinely grant thousands of
> patents on obvious, trivial "inventions" or on things for which
> abundant prior art exists. Companies routinely exploit this sad state
> of affairs to create "landmines" to sabotage legitimate competition,
> as opposed to protecting any substantial investments of their own.
>
> The most important (and by far the most odious) characteristic of a
> patent is that independent reinvention is not a defense to a claim of
> infringement; you can be denied the right to use your own completely
> independent work. The laws of physics and mathematics are the same for
> everyone, and throughout history they (and the inventions that follow
> from them) have often been independently (re)discovered by many people
> at around the same time. For the law to grant exclusive "ownership" of
> laws of nature to those with the best paid lawyers is the height of
> human arrogance. (Yes, I know that in theory, only inventions, not
> physical laws can be patented, and that patent rights are supposed to
> go to the first inventor, not to the one with the most money. But
> the practice is vastly different from the theory.)
>
> I didn't really want to get into a deep discussion of intellectual
> property on this list, but as you can probably tell by now this is one
> of my hot buttons. For more on the trouble with patents, see my web page
>
> http://www.qualcomm.com/people/pkarn/patents.html
>
> --Phil
Phil:
You & P. J. Plauger . . . . now how do we CHANGE this stuff?

Jim

109. [HFSIG:1646] Re: ARRL DCC and the old SS thread

From: "Breon, Roy K" <rkb1@rfpo1.rfc.comm.harris.com>
Date: Wed, 16 Oct 96 09:26:00 DST
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <3264E122@smtpgate.rfc.comm.harris.com>
Rendered body type: text/plain

On October 11 I posted a note regarding our HF modem technology and
philosophy. Many of you wrote me and asked for the source for the
MIL-STD-188-110A describing the HF waveforms.

Unclassified military specifications, military standards, and many other
"public" documents can be obtained from :

Defense Printing Service
700 Robbins Avenue
Philadelphia, PA 19111-5094
215-697-2626

When a specification is issued, it has a revision level letter. Small or
proposed changes are usually attached by means of Change Notices. At some
point, someone decides to issue a new revision incorporating the accumulated
Change Notices. You usually want to ask for the latest revision of the
documents with any Change Notices that may be outstanding.

Hope this helps.

Roy K. Breon
Manager, Marketing
Harris Corporation

RF Communications Division

110. RE: [HFSIG:1656] Re: ARRL DCC and the old SS thread

From: "Breon, Roy K" <rkb1@rfpo1.rfc.comm.harris.com>
Date: Wed, 16 Oct 96 13:57:00 DST
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <32652126@smtpgate.rfc.comm.harris.com>
Rendered body type: text/plain

Phil et al ---

I don't disagree with what you said. I mentioned Eb/N0 but was giving a more simplistic comment in the paragraph on which you commented. Walt Dubose has stated what I was referring to. In other words, the signal would be below the noise level when a 3 kHz receive bandwidth is used. In most situations, with most radios, the operating bandwidth of the radios is fixed at a nominal 3 kHz bandwidth. For the most versatile use of our modems, we stick with that bandwidth and just select various waveforms and data rates , either manually or automatically, to fill it up. In some very automated systems, receive and transmitted bandwidth can be changed on the fly. We have also done special systems where the HF bandwidth is 1 MHz. In these systems you can either have a very high data rate or use processing gain to really pull the signal out of the noise and/or jamming.

Also, as I stated in my post, we consider some modes of the modem to be a spread spectrum type waveform because the full 3 kHz bandwidth would not be necessary to transmit that data rate if standard FSK were used. I believe I stated that a slower data rate was used. As you are probably aware, we also use various coding, interleaving, and in band diversity techniques the choice of which depends upon many factors.

By the way, since you mention Claude Shannon, he was on my Doctoral oral exam committee at MIT.

Also, I always wondered who bought that bridge. Now I know that you "have " it. That must be an interesting off topic story in itself   ;-)

Roy Breon
Harris Corp.
RF Communications division

From: Phil Karn
To: hfsig@tapr.org
Subject: [HFSIG:1656] Re: ARRL DCC and the old SS thread
Date: Tuesday, October 15, 1996 8:19PM

<snipped>

This statement is either wrong or meaningless. What does "negative signal to noise ratio" mean? If you mean Eb/N0 ratios below -1.6 dB, I have this bridge in New York you might be interested in buying. Rumor has it that it's going to be named after Claude Shannon.. :-)

But if you mean a negative signal-to-noise ratio in the operating bandwidth, then your statement is meaningless without specifying both the operating bandwidth and the user data rate. It's easy to make a modem that operates at "negative signal-to-noise ratios" when the bandwidth is much higher than the data rate -- spread spectrum and low rate FEC are two well-known ways to do this.

That's why it's much better to just drop the notion of "signal to noise ratio" when discussing modem performance and talk strictly about Eb/N0 ratios, because they quantify the performance of a demodulator in a way that's independent of bandwidth and data rate.

<snipped>

111. New Viterbi decoder release

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Thu, 17 Oct 1996 01:52:08 -0700 (PDT)
To: tcp-group@ucsd.edu, hfsig@tapr.org
Message-ID: <199610170852.BAA05073@unix.ka9q.ampr.org>
Rendered body type: text/plain

Hi. For anyone interested I have released a new version 2.0 of my Viterbi decoder routines. This release is substantially faster (50%) than my previous version (1.1). The gains came mainly from a change in

the way that final decisions are made on the decoded data bits (I switched from the "register exchange" method to the "traceback" method) and also from continued groveling over the code. Thanks to the longer path memories that are practical with the traceback scheme, BER performance is also slightly improved, though the difference is probably significant only when decoding a high rate punctured code.

The decoding speed on a 133 MHz Pentium is about 259 kilobits/sec for the NASA standard rate 1/2 K=7 convolutional code.

Two versions of the decoder are provided. The first operates on finite-size 'tailed' packets as before. The second is written as a UNIX filter and can operate on a continuous stream of data, closely emulating a hardware Viterbi decoder.

I've updated my ham radio web page with pointers to the new distributions:

http://www.qualcomm.com/people/pkarn/ham.html

--Phil

112. Re: [HFSIG:1649] Re: Why things are the way they are

From: RLANIER@mailb.harris.com (RLANIER)
Date: Wed, 16 Oct 1996 13:08:26 -0400
To: hfsig@tapr.org
Message-ID: <26516550@mailb.harris.com>
Rendered body type: text/plain

>My problem is most definitely NOT with the fact that commercial
>amateur products cost money.

>My problem *is* that they come so often as black boxes. Sure, they
>generally work as advertised. But they generally fail to perform a
>more important function, which is to give the amateur who buys them
>the opportunity to learn as much as he wants about how the device
>functions.

I still cannot understand your rational for insisting that companies supply source code and/or schematics. Should Sun supply the source code for their operating systems? Of course not. Why? Because if they did, hackers and the like could have a working system WITHOUT ever paying Sun one penny.

The amateur market is no different. Sure we want to encourage experimentation. Join a club, find out those involved in this area, read the books supplied by ARRL and others and if that isn't enough, there is plenty of additional material in electrical engineering books. But don't insist that companies give away the very thing they have put alot of effort and money into.

> ...amateur service is supposed to be all about -- self-motivated,
>hands-on, technical education and training.

This is only one facet of ham radio. Some just want to go for the next award or "chew the fat." And others, like us, want to explore the technical aspects and develop the next-generation radios.

> ...cell phones and other commercial systems have made ham radio all
>but irrelevant for emergency communications, and the Internet has made
>ham radio essentially irrelevant in the "international good will"
>category.

I don't see how cell phones have made amateur radio obselete. They still cost money for the phones and even more for the service. A simple CW transceiver running into a dipole is about as simple as it gets and this is also the cheapest. And if your talking to someone, at least you can hear them. With a computer, you can't.

73s de
Tony, KE4ATO

113. [HFSIG:1653] Re: SS thread

From: Robert.Glassey@nmp.nokia.com
Date: Wed, 16 Oct 96 13:18:42 +0100
To: hfsig@tapr.org

Message-ID: <H00002920292f5f8@MHS>
Rendered body type: text/plain

> Why not follow the existing amateur rules that all ready exist? We
> shouldn't cause harmfull interference, period. Neither should someone

> running a SSB system at 100 watts. If we follow good design practices

> and workmanship, which all homebrewers should, then we shouldn't have

> any problems.

One of the first rules that has worked for non-speading modes, is to keep your transmissions within a certian bandwidth. Narrow band receivers are designed to cope with bandwidth limited interference. Once you set this rule asside, your in a new realm of what will and will not cause interference. The old rules dont work for SS, we must consider what the new rules should be.

Its not just a case of quality of equipemnt, no doubt you will endevour to make high quality equipment, its a question of inter-mode compatibility.

> I think what would work even better is if a club or group would
> conduct tests of SS and "regular" systems, side-by-side, and listen
> for interference. What better way than to demonstrate it?

I'm sure it will help, but it will leave many questions unanswered. There are as many variations as there are people to implement them. Some will work others wont. We need some way to sorting out the good from the bad and having something to aim for to ensure we are not wasting our efforts.

> Of course. You always want to start small, test your idea and go

Yip, I had a slightly different meaning in mind. I was thinking of starting with systems that we can be certian will not interfere even if they are quite restrictive. As we gain more experiance we can remove the restrictions as we discover what the technical aspects are. Low power is a start, directional antennas would be a good idea, high procesing gains would help, high frequencies would help. Let go with the most compatible system we can think of, then work out the tradeoffs between say frequency and data rate or whatever.

Cheers,

Rob

114. Re: [HFSIG:1661] Re: Why things are the way they are

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 17 Oct 1996 11:54:33 -0500 (CDT)
To: hfsig@tapr.org
Cc: wa3pay-kc5bji@stic.net (Teri Thomas), thomas@uthscsa.edu (Charles Thomas),
choffman@pelican.davlin.net (Ric Hoffman),
102651.2105@compuserve.com (Al Uvietta),
valdez1@samhou-basops.army.mil (Ed Valdez)
Message-ID: <199610171654.LAA18556@www.kelly-afb.org>
Rendered body type: text/plain

In Tony's message he writes:
[stuff deleted]
> > ...cell phones and other commercial systems have made ham radio all
> >but irrelevant for emergency communications, and the Internet has made
> >ham radio essentially irrelevant in the "international good will"
> >category.
>
> I don't see how cell phones have made amateur radio obselete. They
> still cost money for the phones and even more for the service. A
> simple CW transceiver running into a dipole is about as simple as it
> gets and this is also the cheapest. And if your talking to someone, at
> least you can hear them. With a computer, you can't.
>
>
>
> 73s de
> Tony, KE4ATO
>

I have to agree with Tony.

Cellphone "cells" are the first communications media to become
overloaded and usless in an emergency.

Also, we have pointed out to much of the "community" that
cellphones can not run "net" operations and the "community" agrees.

The three major radio clubs in San Antonio and the City CD,
over the last couple of months, have averaged 1.5 communications
events per week. This is providing communications for "community"
events that they cannot get by using cellphones, trunking radio
systems or GMRS.

"Nets" are the most valuable communications asset during an emergency.
Local and state governments _CANNOT_ maintain a full time cadre of
paid and trained radio operators, with equipment...taxpayers will
never allow them to do this. Thus, the trained hamradio operator
with his equipment becomes a valuable asset to local and state
governments during emergencies.

Walt

| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development. |

	The greatest dangers to liberty
Walt DuBose - K5YFW	lurk in insidious encroachment
E-Mail k5yfw@www.kelly-afb.org	by men of zeal, well-meaning
Business Telephone: (210)925-6081	but without understanding.
Home Telephone: (210)696-3196	
	- Justice Louis D. Brandeis

115. Re: [HFSIG:1661] Re: Why things are the way they are

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 17 Oct 1996 16:32:08 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610172332.QAA18104@servo.qualcomm.com>
Rendered body type: text/plain

I still cannot understand your rational for insisting that companies
     supply source code and/or schematics. Should Sun supply the source
     code for their operating systems? Of course not. Why? Because if they
     did, hackers and the like could have a working system WITHOUT ever
     paying Sun one penny.

Again, copyrights are designed to deal with that issue, though I don't
want to veer off again into a general discussion on intellectual
property.

Your choice of example helps make my original point. Precisely because
of the proprietary nature of most commercial operating systems,
efforts such as FreeBSD, NetBSD and Linux have sprung up to make a
nonproprietary operating system, with full source code, readily
available to anyone. And the GNU project has succeeded in producing a
large collection of utilities under the same terms.

In many cases the freely available software is as good as or even
better than the proprietary stuff. This is partly because a suitably
motivated and talented software volunteer can often outperform a paid
programmer who is just doing it for the money, and also because the
widespread availability of the software creates a much larger pool of
programming talent that can contribute to the project.

I did *not* say that amateur equipment manufacturers should be
required to disclose their source code. That is their right. I was
merely lamenting the fact that they have chosen not to do so. This
creates an opportunity for an independent volunteer amateur effort
very similar to the Linux and GNU projects to fill the gaps left by
the increasingly proprietary commercial vendors, and perhaps to
outperform them.

Phil

116. Re: [HFSIG:1661] Re: Why things are the way they are

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 17 Oct 1996 16:43:27 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610172343.QAA18128@servo.qualcomm.com>
Rendered body type: text/plain

> ...amateur service is supposed to be all about -- self-motivated,
>hands-on, technical education and training.

     This is only one facet of ham radio. Some just want to go for the next
     award or "chew the fat." And others, like us, want to explore the
     technical aspects and develop the next-generation radios.

As I've said many times, I have no problem with those who are simply out
to enjoy themselves. A volunteer service wouldn't last long if you banned
having fun, because what other reason would there be to participate?

My problem is with the fact that hams make up only about 0.5% of the
US population, so if we're to keep some increasingly valuable spectrum
to ourselves we have no choice but to sell ourselves to the remaining
99.5% on the basis of something other than how much fun the 0.5% is
having. That means public service, education, etc.

     I don't see how cell phones have made amateur radio obselete. They
     still cost money for the phones and even more for the service. A
     simple CW transceiver running into a dipole is about as simple as it
     gets and this is also the cheapest. And if your talking to someone, at
     least you can hear them. With a computer, you can't.

I could repeat the story of how I finally came to buy a cell phone,
but you've probably heard it before. Suffice to say I had an extremely
frustrating experience with using ham radio to summon help in a road
emergency (serious injury) situation when a cell phone would have made
a direct 911 call easy. As for cost, portable cell phones are now
substantially cheaper than their nearest ham equivalent (the 2m HT),
and service rates are quite low if you rarely make calls. (Even an
unregistered phone can still make 911 calls.)

CW is completely irrelevant for any credible modern emergency situation.

Phil

117. Re: [HFSIG:1659] Re: ARRL DCC and the old SS thread

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 17 Oct 1996 16:49:46 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610172349.QAA18141@servo.qualcomm.com>
Rendered body type: text/plain

>I don't disagree with what you said. I mentioned Eb/N0 but was giving a
>more simplistic comment in the paragraph on which you commented. Walt

Right, I was just trying to inject some rigor into what had started to
sound like a Harris marketing blurb. :-)

>Also, as I stated in my post, we consider some modes of the modem to be a
>spread spectrum type waveform because the full 3 kHz bandwidth would not be
>necessary to transmit that data rate if standard FSK were used. I believe I

Indeed. I've made this exact point for some time to show that we hams
can use the SSB-bandwidth equipment we already have to experiment with
spread-spectrum techniques on a "scaled down" basis. All the SS stuff
would be done purely in software, which is ideal for inexpensive
experimentation.

Phil

118. Re: [HFSIG:1663] Re: Why things are the way they are

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 17 Oct 1996 16:59:56 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610172359.QAA18150@servo.qualcomm.com>
Rendered body type: text/plain

Cellphone "cells" are the first communications media to become
        overloaded and usless in an emergency.

I wish I had a nickle for each time I've heard that said. If our only
advantage vs cellular telephony in an emergency are our generous

spectrum allocations and tiny numbers, then we're in serious trouble
as the obvious fix is to reallocate our spectrum to cellular telephony
where it can be used more efficiently.

Yet I dispute the actual claim. There are many more audio channels in
my one local cell site than are available in all the ham repeaters in
town. And cellular system capacity is expanding so rapidly as to make
anecdotes even about relatively recent events (like the Loma Prieta
and Northridge earthquakes) rather meaningless as future predictions.

If I needed to make a phone call in an emergency I think I'd get
through faster by just pressing the SND key until I got through than
to wait for the local repeater to clear to use the autopatch.

>"Nets" are the most valuable communications asset during an emergency.

Assuming that is true (which I don't -- nets and point-to-point mode
operation each has its place), it is not true that net-style operation
is unique to amateur radio. Trunking systems are already widespread.
Adding net-style operation to CDMA digital cellular has been an
ongoing government-sponsored project at Qualcomm for several years.

Phil

119. Re: [HFSIG:1664] Re: Why things are the way they are

From: Jim Sanford <jsanford@infi.net>
Date: Thu, 17 Oct 1996 21:36:45 -0400
To: hfsig@tapr.org
Cc: w4hfz@amsat.org, ko4lw@amsat.org
Message-ID: <3266DF2D.3F93@infi.net>
Rendered body type: text/plain

Phil Karn wrote:
>
>   I still cannot understand your rational for insisting that companies
>   supply source code and/or schematics. Should Sun supply the source
>   code for their operating systems? Of course not. Why? Because if they
>   did, hackers and the like could have a working system WITHOUT ever
>   paying Sun one penny.
>
> Again, copyrights are designed to deal with that issue, though I don't
> want to veer off again into a general discussion on intellectual
> property.
>
> Your choice of example helps make my original point. Precisely because
> of the proprietary nature of most commercial operating systems,
> efforts such as FreeBSD, NetBSD and Linux have sprung up to make a
> nonproprietary operating system, with full source code, readily
> available to anyone. And the GNU project has succeeded in producing a
> large collection of utilities under the same terms.
>
> In many cases the freely available software is as good as or even
> better than the proprietary stuff. This is partly because a suitably
> motivated and talented software volunteer can often outperform a paid
> programmer who is just doing it for the money, and also because the
> widespread availability of the software creates a much larger pool of
> programming talent that can contribute to the project.
>
> I did *not* say that amateur equipment manufacturers should be
> required to disclose their source code. That is their right. I was
> merely lamenting the fact that they have chosen not to do so. This
> creates an opportunity for an independent volunteer amateur effort
> very similar to the Linux and GNU projects to fill the gaps left by
> the increasingly proprietary commercial vendors, and perhaps to
> outperform them.
>
> Phil
Since I either tossed the match or initial gallon of gasoline on this
fire, it's time for me to throw another . . .

I do not expect radio manufacturers to publish their source code.
Although a lot could be improved if they did, just look at the
tremendous improvement Chipswitch made to the Uniden radios --what a
great IF rig!
Look at the problems the Ubiquitous FT736 has -- a computer command set
which is practically one-way, vulnerable to spurious transmitter
lock-ups, and basically primitive. A little cooperation on Yaesu's part
could allow someone to improve it (a la Chipswitch/Uniden) improve a

gross deficiency in a pretty good radio, and all at ZERO cost to Yaesu.
Of course the consequence could be even MORE 736's sold . . .

I do, however, _DEMAND_!!!! that anyone putting a waveform on the ham
bands publish the details of it, so that it can be understood, and
perhaps duplicated in DSP by other folks. Proprietary waveforms on the
ham bands is inexcusable, unacceptable, and should be discouraged by
hams taking their checkbook elsewhere. Henry was trying hard to sell
Clover to the Military (at least from all the stuff I saw as a Naval
officer), and when other, better systems made them non-players, they
tried to recover their costs from paying hams -- shades of Bill Gates.
I notice that they're finally dropping their prices . . . . Clover may
be a good system (althought the military and NATO didn't think so), but
until there's sufficient information in the PUBLIC DOMAIN, it should be
ignored.

Phil and Johan (and others) are doing enough with open algorithms to
enhance HF digital communications that we don't need to patronize or
subsidize proprietary waveforms. The fact that they choose to publish
their code is merely icing on the cake.

So, in summary: If you've got a proprietary waveform, take it out of
the ham bands -- go find a gullible congressman, general, or admiral to
buy it. Hams, if you want to push the state of the art on HF digital
communications, keep your $$ away from the proprietary stuff, and go
with the guys who are CONTRIBUTING SIGNIFICANTLY TO THE STATE OF THE ART
-- part of what Ham Radio is all about, STILL!

Feel free to flame on -- I've been screamed at by ADM Rickover, so have
a ball. I'm willing to listen to any reason why Phil or Johan et al
might choose to keep their code private, but inflexible on the
proprietary waveform issue.

73, Jim
WB4GCS@amsat.org

120. Wanted: SIG Chair

From: Greg Jones <wd5ivd@tapr.org>
Date: Fri, 18 Oct 1996 00:11:36 -0500 (CDT)
To: hfsig@tapr.org (HF SIG mailing)
Message-ID: <199610180511.AAA12704@tapr.org>
Rendered body type: text/plain

With all the red hot debate, maybe this is the time to say that this SIG
needs a Chair to replace Johan.

Johan asked to be replaced at the ARRL and TAPR DCC in Seattle.

Basically, Johan has other outside commitments that are requiring more of
his time and he would like to see some one fresh take over the SIG. Johan
is the founder of this SIG and both he and I would really like to see the
direction he began continue. As Johan stated at the DCC, he is not going
away, he just needs someone to take over the position of chair.

Responsibilities are simple:

1. Watch over this list. Which means try to help focus technical discussion
and reply to those that send unsub and sub messages to the list :-)

2. Write a short overview of SIG activities twice a year for the TAPR PSR

3. Help organize the HF SIG meeting at the ARRL and TAPR DCC

4. Organize or assign some one to handle the HF SIG ftp area.

5. Other tasks as assigned by the SIG chair them self.

Those are the basics. Johan has been able to go way beyond these simple
tasks, but anyone starting now would only need to do these. These simple
tasks allow the SIG to run in an orderly fashion and provide more than just
a mailing list on HF digital topics.

I believe some of the key elements required to be a good SIG chair include
looking for connections to other areas, forwarding messages of interest to
the PSR editor for potential articles, and trying to look for concepts and
projects that can be moved from the SIG to a much wider audience.

I'll let Johan add additional comments.

This is the time to find a new chair and someone must rise to the task of being chair.

If you think you might be interested, please contact Johan or myself.

Cheers - Greg, WD5IVD
Pres TAPR

121. Re: [HFSIG:1668] Re: Why things are the way they are

From: Rolf Sommerhalder <rs@ife.ee.ethz.ch>
Date: Fri, 18 Oct 1996 09:50:32 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199610180750.JAA19000@gillespie.ee.ethz.ch>
Rendered body type: text/plain

Jim wrote:
> ...
> I do, however, _DEMAND_!!!! that anyone putting a waveform on the ham
> bands publish the details of it, so that it can be understood, and
> perhaps duplicated in DSP by other folks. Proprietary waveforms on the
> ham bands is inexcusable, unacceptable, and should be discouraged by
> hams taking their checkbook elsewhere.
> ...
> Phil and Johan (and others) are doing enough with open algorithms to
> enhance HF digital communications that we don't need to patronize or
> subsidize proprietary waveforms. The fact that they choose to publish
> their code is merely icing on the cake.
> ...

I am thinking roughly along the same lines that _specifications_ should be laid open, even though _implementations_ may be kept as a trade secret by the vendors (in the HAM as well as in commercial markets). Of course, the specifications must be precise enough, so that other implementors can produce compatible, i.e. fully interopable, devices.

This is the approach taken by ITU (and many other standardisation bodies as well) in its recommendations, where in V.42bis for example, only the data compression algorithms are specified precisely, but no implementation details are provided.

I have been discussing this issue with the holder of the rights on Pactor (-1/-2) for commercial markets some two years ago, and I was unable to convince him that he could create a bigger market for Pactor-2 if its specification would be open. Commercial buyers too are reluctant buying products which come from a single manufacturer (what if he goes out of business ? etc.).
I understand that it is tough for the inventors and their investors to disclose some of their key know-how considering that they want to recoup their investments and make some profit for the risks they have taken. In the longer term, I believe, they could do that also in a more open environment, provided of course that they stay in business by delivering good products to an expanding market, and that they continue to lead innovation/evolution. They choose another way, so far.

I guess that the IS-95/CDMA cellular phone standard is an interesting case which maybe in support of my ideas. Phil might want expand on similar mechanisms which are applied by his employer to create a new market for CDMA (which did not even exist before), and to compete against established GSM technology.

73,
Rolf, HB9CWP

122. Re: [HFSIG:1665] Re: Why things are the way they are

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Fri, 18 Oct 1996 09:54:20 +0200
To: hfsig@tapr.org
Message-ID: <326737AC.877@ife.ee.ethz.ch>
Rendered body type: text/plain

Phil Karn wrote:

> CW is completely irrelevant for any credible modern emergency situation.
Since ID4 we all know that only CW can save the world :-)

Tom

123. Re: [HFSIG:1671] Re: Why things are the way they are

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 18 Oct 1996 03:46:52 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610181046.DAA19284@servo.qualcomm.com>
Rendered body type: text/plain

>> CW is completely irrelevant for any credible modern emergency situation.
>Since ID4 we all know that only CW can save the world :-)

I laughed out loud at several points in that movie when I was probably
not supposed to. That was one of them.

It would have been much better had the governments of earth used PGP
and the Internet (both being widely available around the globe) to
coordinate the counterattack. Maybe the aliens wouldn't have figured
it out quite as quickly. :-)

Phil

124. Re: [HFSIG:1668] Re: Why things are the way they are

From: RLANIER@mailb.harris.com (RLANIER)
Date: Fri, 18 Oct 1996 09:18:35 -0400
To: hfsig@tapr.org
Message-ID: <267837a0@mailb.harris.com>
Rendered body type: text/plain

>Feel free to flame on -- I've been screamed at by ADM Rickover, so
>have a ball. I'm willing to listen to any reason why Phil or Johan et
>al might choose to keep their code private, but inflexible on the
>proprietary waveform issue.

>73, Jim
>WB4GCS@amsat.org

Look guys, this is getting out of hand!!! If you want to get "into
it," please do so off of this list, such as Phil and I have.

I want to clairify, ONCE AND FOR ALL, my position. I think it is great
that hams such as Phil, Johan, et al, are working to push the
state-of-the-art in ham radio. PLEASE KEEP DOING IT!!! I for one spend
as much time as I can doing the same thing. And as I have stated
before, I will share my results.

We do so because we like to, because it is FUN! But as I have said
before, a company, ANY COMPANY, exists to make money, whether it is to
provide a service or to sell a product. And in amateur radio, there is
plenty of competition, especially between the "big guns," Yaesu, Icom
and Kenwood. These companies work hard to come up with the latest and
greatest, so they can make a profit. If Kenwood did some research on
the merits of using DSPs in the IF stage and then published it,
including source code, Yaesu or Icom could use it to make their next
product line. Kenwood just contracted out work for FREE! That is not
going to cut it in the business world guys, no matter how much you
complain!

Start your own company and rely SOLELY of the profits of that company
and I guarantee you will change your mind.

73s de
Tony, KE4ATO

125. Re: [HFSIG:1652] Ham Radio Clip Art

From: "Rick Williams" <rwilliam@cesa4.k12.wi.us>
Date: Fri, 18 Oct 1996 15:56:12 -0500
To: <hfsig@tapr.org>
Message-ID: <199610182114.QAA482952@gomer.wiscnet.net>
Rendered body type: text/plain

Larry Bush, W5NCD had said:
> Does anyone know of a source of Ham Radio Clip Art?
> I Want to make up my own QSL and calling Cards.
>

I made up a simple QSL format using a word processor and stock clip art. I
used a globe clip art pattern and a table format using MS Word and make

four duplicates per laser printed page. I can either print small amounts if I have a need for a few special cards or merely substitute the call sign and name and address, etc. for different folks.

For example, it was trivial to make up a template for my wife or daughter to use for their cards. Since they almost never need them, it can be run off in quantities of four at a time or could use the master for "printing" in a copy machine. Tried higher weight paper (like about 90# I think) and for the most part got it to go thru the machine but that is pushing the limits.

I will never again purchase QSL cards from a printer. The only way it makes sense is if you need really large quantities, vis a vis contesting, DXpeditioning, or similar multi-thousand quantities, or want multi-color or picture types with special effects.

Rick, KV9U

126. PC-DSP and PC-SIM Order

From: "James R. Kauten, MD" <kauten@atl.mindspring.com>
Date: Tue, 22 Oct 1996 21:00:03 -0400
To: arne_luehrs@grenoble.hp.com, n0aot@amsat.org, rsmith@internetmci.com, bgaf@annap.infi.net, mullaneb@mccc.edu, xbrucex@popmail.mcs.net, crlbyers@garlic.com, dsp-93@tapr.org, ebossa@acm.org, 101365.1113@compuserve.com, wd5ivd@tapr.org, hbs@crl.com, hfsig@tapr.org, govier02@sarenet.es, k4gvo@america.net, jsanford@infi.net, hl1sqi@neptune.samwoo.co.kr, kgoodwin@draper.com, kbanke@mage.qualcomm.com, johnson@mmsi.com, 100577.1452@compuserve.com, MS16948@LTU.EDU, rsuban@promitheus.maltanet.omnes.net, rbiasiot@nji.com, schultew@uni-duesseldorf.de, tedm@voyager.co.nz, wb8oue@amsat.org, LANIER.R.A-@smtpgty.bwi.wec.com, twatson@magic1.org, wpreez@csir.co.za, ylshih@alumni.caltech.edu
Message-ID: <1.5.4.32.19961023010003.006d3abc@pop.atl.mindspring.com>
Rendered body type: text/plain

As you are aware, many folks expressed an interest in the software: PC-DSP and PC-SIM by DSP Solutions (http://www.dspsolutions.com). TAPR also has a description of the software on their web page (http://www.tapr.org). In the most recent PSR-TAPR newsletter there is also a description. If there are any other folks interested in purchasing the software other than those of us who have placed our orders, place your order thru TAPR so that a final order can be sent in. We need a minimum order of 21 before an order can be placed.

Thank you.

Jim Kauten, MD, KO4RQ
=====
James R. Kauten, M.D., KO4RQ (kauten@mindspring.com)
=====
Phone: (404) 355-9537 (Fax) (404) 355-9515 (Daytime)
Address: James R. Kauten, M.D., 95 Collier Road, Suite 2055
Atlanta, GA, 30305-1955
=====

127. Re: [HFSIG:1670] Re: Why things are the way they are

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Wed, 23 Oct 1996 02:28:53 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610230928.CAA03874@unix.ka9q.ampr.org>
Rendered body type: text/plain

>I guess that the IS-95/CDMA cellular phone standard is an interesting
>case which maybe in support of my ideas. Phil might want expand on similar
>mechanisms which are applied by his employer to create a new market for CDMA
>(which did not even exist before), and to compete against established GSM
>technology.

The IS-95 CDMA waveforms are described in exacting detail in a public document with the number, coincidentally enough, "IS-95". :-)

The technology is, however, heavily laced with patents which are sadly unavoidable in a purely commercial system. I am hoping against hope that we can avoid these in an open waveform design intended for amateur use.

Phil

128. Re: [HFSIG:1582] Re: Development Environment

From: N0AOT@aol.com
Date: Wed, 23 Oct 1996 10:50:46 -0400
To: hfsig@tapr.org
Message-ID: <961023105045_216662458@emout08.mail.aol.com>
Rendered body type: text/plain

Hi All--

Suggestion: If phase reference and clock information are really the critical part of all this, why not take the easy way out and use a world-wide external reference for the clock information, such as could be derived from a GPS signal?

Tom Clark has already done most of the work on this with his Totally Accurate Clock project. It would mean that a modem would have to incorporate a GPS engine, but what they heck, they are getting cheap and it sure seems like it would solve lots of implementation problems!

--Bob (n0aot@amsat.org)

>>The real trick is in deriving the incoming carrier phase reference.
>>And doing this well is a real bitch (if not downright impossible) at
>>low data rates on channels with the slightest bit of unpredictable
>>movement (satellite or ionospheric motion, etc).
>
>Yes, precisely. THe "block" on the "simulator" just used the clock from the
>transmit side and didn't bother to derive *any* clocks. It was OK for
>evaluating some things, but not nearly as useful as its price tag would have
>indicated...
>
>Cheers,
>
>Lyle
>
>

129. Re: [DSP-93:2340] EVM56002 radio interface schematic?

From: "Steven R. Bible" <srbible@gnatnet.net>
Date: Wed, 23 Oct 1996 18:27:29 -0400
To: dsp-93@tapr.org, hfsig@tapr.org
Message-ID: <1.5.4.32.19961023222729.0067c0e0@gnatnet.net>
Rendered body type: text/plain

Mark, I am way overdue in getting the DSP Experimenters Web Pages on the TAPR Web site updated and improved. I just happen to be working on a DSP56002EVM page. It will have the schematic and PCB layouts available in both PDF and PS format. Look for the page at

   http://www.tapr.org/dsp/evm56k

As always, please let me know of any suggestions and comments to improve the pages.

73, Steve N7HPR

At 09:11 AM 10/23/96 -0500, you wrote:
>Hello all,
>
>In anticipation of the EVM56002 arrival, I was considering the radio
>interface issue. Johan, KC7WW, has a schematic of such a beast, but I
>cannot read the file types. He has posted both the following:
>=======
>Native EVM56002 Applications (KC7WW)
>
>EVMPCB.PLT
>   KC7WW's EVM radio interface schematic in HPGL format.
>EVMPCB.SCH
>   KC7WW's EVM radio interface schematic in ORCAD format.
>
>========
>
>Can anybody on this list make some sort of translation of these into either
>.PDF or .BMP file types so some of us nontechnical types can print the
>schematic? The files are available at:
>
>ftp://ftp.tapr.org/tapr/SIG/hfsig/upload/evm56k3.zip
>

>
>Thanks,
>
>-------------------------
>Mark L. Hammond [KC4EBR]
>Campbell University
>
>
>

 - Steve, N7HPR
 (n7hpr@tapr.org)

130. Linux and Clover?

From: "Michael A. Burke" <mburke@llajta.nrc.bolnet.bo>
Date: Thu, 24 Oct 1996 01:25:43 -0400
To: hfsig@tapr.org
Cc: bbssig@tapr.org
Message-ID: <199610232130.BAA24970@llajta.nrc.bolnet.bo>
Rendered body type: text/plain

Anyone out there know of any drivers or software which works with the
HAL Clover boards under Linux?

Mike - cp5vw

... Mike

131. Re: [HFSIG:1677] Re: Development Environment

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 24 Oct 1996 02:33:39 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610240933.CAA17287@servo.qualcomm.com>
Rendered body type: text/plain

>Suggestion: If phase reference and clock information are really the critical
>part of all this, why not take the easy way out and use a world-wide external
>reference for the clock information, such as could be derived from a GPS
>signal?

Because the round trip path delay is not known precisely enough, and
it also varies. External GPS-derived timing may be good enough for
symbol timing, if the symbol rate is not too fast and if the
propagation delay can be estimated. It won't work for carrier phase,
though. Even on HF the wavelength is much smaller than the delay
uncertainty.

This is not to say that GPS isn't quite useful as a stable frequency
reference to eliminate one of the sources of uncertainty in the system.
But phase tracking in the modem itself still seems essential.

Phil

132. GPS for clock recovery

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Thu, 24 Oct 1996 12:56:33 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <199610241956.AA277236993@polecat.sr.hp.com>
Rendered body type: text/plain

Phil Karn <karn@qualcomm.com> wrote:

>>Suggestion: If phase reference and clock information are really the critical
>>part of all this, why not take the easy way out and use a world-wide external
>>reference for the clock information, such as could be derived from a GPS
>>signal?
>
>Because the round trip path delay is not known precisely enough, and
>it also varies.

The path delay between satellite and GPS receiver is compensated for by
the clever algorithms in the GPS unit (using timing data from several
satellites.) The delay between receiver and transmitter might be a
problem, however.

>External GPS-derived timing may be good enough for
>symbol timing, if the symbol rate is not too fast and if the
>propagation delay can be estimated.

Seems like the GPS timing accuracy should be fine. If the GPS receiver
can give position to within 100 meters, I assume it must be capable of
timing accuracy on the order of 100/c = 333 ns, which would be plenty
good enough for the symbol rates used on HF.

>It won't work for carrier phase,
>though. Even on HF the wavelength is much smaller than the delay
>uncertainty.

Yes, the delay uncertainty between the HF transmitter and receiver is
not generally known and varies (usually slowly) from time to time with
changes in propagation conditions. But it seems like there ought to be
a way around that problem without having to do a full-fledged clock
recovery in the demodulator.

>This is not to say that GPS isn't quite useful as a stable frequency
>reference to eliminate one of the sources of uncertainty in the system.
>But phase tracking in the modem itself still seems essential.

I know that Tom Clark W3IWI of AMSAT (and NASA) has been working on these
types of questions. His GPS-based "TAC" Totally Accurate Clock (or if
you prefer, Tom A. Clark) unit can give very accurate timing information.
I emailed him to see if he has any thoughts on the subject.

Alan Bloom N1AL

133. Re: [HFSIG:1681] GPS for clock recovery

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 24 Oct 1996 14:45:20 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199610242145.OAA03883@servo.qualcomm.com>
Rendered body type: text/plain

>The path delay between satellite and GPS receiver is compensated for by
>the clever algorithms in the GPS unit (using timing data from several
>satellites.) The delay between receiver and transmitter might be a
>problem, however.

Right. Actually the GPS receiver solves a set of simultaneous equations
that solve for time along with location. It's easy to get time accurate
to a microsecond or even considerably less out of an inexpensive receiver.

But as I said, that's not the problem in using GPS as a carrier phase
and/or symbol timing reference in a HF communication system. The round
trip delay simply isn't known accurately enough -- you don't
necessarily know in advance where the other station is, and the
constantly shifting ionosphere makes the delay hard to estimate even
if you do know where the other station is. And in the case of carrier
phase via the ionosphere, it never stays constant.

Good modulation schemes exist that do not require precise knowledge of
carrier phase, such as FSK (all forms) and differentially coherent PSK.
Symbol timing is not that hard to extract from the incoming signal at the
rates typically used on HF. While external reference timing might be
useful at very slow symbol rates, the ionospheric doppler spread sets
a lower limit on symbol rate that makes this less than highly useful on HF.

The main application for GPS symbol synchronization that makes sense
is QRP EME, where the symbol rates are slow enough for this to work
and the round trip distance to the moon can be accurately calibrated
(assuming you know the location of the other station).

Phil

134. TAPR DSP56002EVM Web Pages

From: "Steven R. Bible" <srbible@gnatnet.net>
Date: Sat, 26 Oct 1996 08:36:20 -0400
To: dsp93@tapr.org, hfsig@tapr.org
Message-ID: <1.5.4.32.19961026123620.0075ff18@gnatnet.net>
Rendered body type: text/plain

Hi All,

I've spent the day refining and adding to the DSP56002EVM web page on the
TAPR Web server.

    http://www.tapr.org/dsp/evm56k/index.html

I've managed to catalog all of the experimenting that has been done on the
EVM. There's KC7WW, ZS6AWK, and EA2ARU's work featured. If you know of any
other projects please email me.

With the success of the TAPR group purchase of the EVM, there's going to be
a couple of hundred new experimenters out there. Hopefully, this page can
be a good starting place for these folks and veterans alike.

As always, suggestions are welcome in improving the pages.

73,

 - Steve, N7HPR
 (n7hpr@tapr.org)

135. Re: [HFSIG:1683] TAPR DSP56002EVM Web Pages

From: forrerj@peak.org (Johan Forrer)
Date: Sat, 26 Oct 1996 21:24:18 -0800
To: hfsig@tapr.org
Message-ID: <199610270423.VAA22797@PEAK.ORG>
Rendered body type: text/plain

Hi Steve,

Your efforts are much appreciated.
I may just add that there have been some useful bug fixes and improvements to
the Leonid OS that folks should know about. Some of these were contributed by
Doug Braun; the most notable is a way to load the OS into FLASH so that it
boots
that right away on reset.

When the EVM support group starts up, I'll be happy to post a summary of these
recent additions and improvements.

73's till later

--Johan

P.S. Sorry for not showing much activity lately - things are just hectic.

>Hi All,
>
>I've spent the day refining and adding to the DSP56002EVM web page on the
>TAPR Web server.
>
>   http://www.tapr.org/dsp/evm56k/index.html
>
>I've managed to catalog all of the experimenting that has been done on the
>EVM. There's KC7WW, ZS6AWK, and EA2ARU's work featured. If you know of any
>other projects please email me.
>
>With the success of the TAPR group purchase of the EVM, there's going to be
>a couple of hundred new experimenters out there. Hopefully, this page can
>be a good starting place for these folks and veterans alike.
>
>As always, suggestions are welcome in improving the pages.
>
>73,
>
> - Steve, N7HPR
> (n7hpr@tapr.org)
>
>

136. Re: [HFSIG:1684] Re: TAPR DSP56002EVM Web Pages

From: Greg Jones <wd5ivd@tapr.org>
Date: Sat, 26 Oct 1996 23:54:20 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199610270454.XAA11279@tapr.org>
Rendered body type: text/plain

> When the EVM support group starts up, I'll be happy to post a summary of these
> recent additions and improvements.

Good news! It has :-)

You join the DSP@TAPR.ORG list by sending e-mail to

listserv@tapr.org
in the message put
subscribe dsp Your Name (First and Last, Call please)

You can also use the SIG page in www.tapr.org

We should start shipping the first units out next week. Motorola still owes
us 60+ units to fullfill the order we have here. They are coming last I
heard.

Anyway -- this should be fun :-)

Cheers - Greg, WD5IVD

November 1996

Original source file: hfsig.9611
Source: https://archive.org/details/tapr-1997
Recovered from: https://archive.org/download/TAPR-HF_1996-11/TAPR-HF_1996-11.txt

1. FW: Wavecom W41PC DSP data decoder card

From: "Bloom, Jon, KE3Z" <jbloom@arrl.org>
Date: Fri, 1 Nov 1996 12:38:00 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <m0vJNZc-000f5hC@mgate.arrl.org>
Rendered body type: text/plain

This bounced around the HQ email system until it landed in my inbox.
Might be of interest to some on this group, I thought.

 -- Jon, KE3Z

Date: 31 Oct 96 10:02:26 EST
From: Joerg Klingenfuss <101550.514@CompuServe.COM>
To: BlindCopyReceiver:;@compuserve.com
Subject: Wavecom W41PC DSP data decoder card
Message-ID: <961031150226_101550.514_IHK54-1@CompuServe.COM>

Dear friends

we would like to draw your attention to the brandnew Wavecom W41PC DSP
data
analyzer and decoder card for Windows 95 PC systems. This professional
equipment
is now available and it is really the ultimate decoder. We will start
distribution in early December and our "waiting list" is now open. For a
detailed description and some sample colour screenshots please refer to
http://ourworld.compuserve.com/homepages/Klingenfuss/ . A free colour
brochure
will be available soon. Thank you for your interest, and best wishes for
the
main listening season!

73 es best DX, Joerg Klingenfuss

Klingenfuss Radio Monitoring
Hagenloher Str. 14
D-72070 Tuebingen
Germany
Phone ++49 7071 62830
Fax ++49 7071 600849
E-Mail 101550.514@compuserve.com
WWW-Site http://ourworld.compuserve.com/homepages/Klingenfuss/

2. DSP/Modem article

From: forrerj@peak.org (Johan Forrer)
Date: Sun, 03 Nov 1996 19:06:41 -0800
To: HFSIG@tapr.org
Message-ID: <199611040305.TAA17295@PEAK.ORG>
Rendered body type: text/plain

Hi,

Thought I would mention a recent article in Circuit Cellar INK
(November 1996) that may be of interest.

Page 20; "Real-Time DSP Modems with a PC and Sound Card".

Covers some experimental work on a V.22 modem that can run on a
486 class PC with a sound card. Source code in C and assembly
are included. Interesting article if you have not yet been
introduced to modem theory and techniques.

Page 14; "Bulding a High-Performance DSP System"

Covers the architecture for the EZ-Kit. Quite informative if
you have not used the latest ADSP-21xx family chips. Actually
the chip has a neat DMA channel that is worth something but
not given much attention in the article.

--Johan

3. Re: [HFSIG:1687] DSP/Modem article

From: Jim Sanford <jsanfor@exis.net>
Date: Mon, 04 Nov 1996 06:42:03 -0500
To: hfsig@tapr.org
Message-ID: <327DD68B.36D6@exis.net>
Rendered body type: text/plain

Johan Forrer wrote:
>
> Hi,
>
> Thought I would mention a recent article in Circuit Cellar INK
> (November 1996) that may be of interest.
>
> Page 20; "Real-Time DSP Modems with a PC and Sound Card".
>
> Covers some experimental work on a V.22 modem that can run on a
> 486 class PC with a sound card. Source code in C and assembly
> are included. Interesting article if you have not yet been
> introduced to modem theory and techniques.
>
> Page 14; "Bulding a High-Performance DSP System"
>
> Covers the architecture for the EZ-Kit. Quite informative if
> you have not used the latest ADSP-21xx family chips. Actually
> the chip has a neat DMA channel that is worth something but
> not given much attention in the article.
>
> --Johan
Johan:
Not familiar with this magazine . .. where do I get one? Is
"Circuit Cellar" the name of the mag????

Thanks, Jim
wb4gcs@amsat.org

---

4. Re: [HFSIG:1688] Re: DSP/Modem article

From: rlanier@su102s.ess.harris.com (Tony Lanier)
Date: Mon, 4 Nov 1996 08:42:32 -0500
To: hfsig@tapr.org
Message-ID: <199611041342.IAA17814@Sopchoppy.GASD102designcenter>
Rendered body type: text/plain

> From hfsig@tapr.org Mon Nov 4 08:00:28 1996
> Date: Mon, 4 Nov 1996 05:42:30 -0600 (CST)
> Originator: hfsig@tapr.org
> From: Jim Sanford <jsanfor@exis.net>
> To: hfsig@tapr.org
> Subject: [HFSIG:1688] Re: DSP/Modem article
> X-Listprocessor-Version: 6.0 -- ListProcessor by Anastasios Kotsikonas
> X-Comment: Tucson Amateur Packet Radio HF Special Interest Group
> Content-Transfer-Encoding: 7bit
> Mime-Version: 1.0
>
> Johan Forrer wrote:
> >
> > Hi,
> >
> > Thought I would mention a recent article in Circuit Cellar INK
> > (November 1996) that may be of interest.
> >
> > Page 20; "Real-Time DSP Modems with a PC and Sound Card".
> >
> > Covers some experimental work on a V.22 modem that can run on a
> > 486 class PC with a sound card. Source code in C and assembly
> > are included. Interesting article if you have not yet been
> > introduced to modem theory and techniques.
> >
> > Page 14; "Bulding a High-Performance DSP System"
> >
> > Covers the architecture for the EZ-Kit. Quite informative if
> > you have not used the latest ADSP-21xx family chips. Actually
> > the chip has a neat DMA channel that is worth something but
> > not given much attention in the article.
> >
> > --Johan
> Johan:
> Not familiar with this magazine . .. where do I get one? Is
> "Circuit Cellar" the name of the mag????

>
> Thanks, Jim
> wb4gcs@amsat.org

What issue is it in?

Tony, KE4ATO

5. Re: DSP/Modem article

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Mon, 04 Nov 1996 16:38:43 +0100
To: hfsig@tapr.org
Message-ID: <327E0E03.2783@ife.ee.ethz.ch>
Rendered body type: text/plain

While we're at implementing modems using general purpose
PC's and soundcards, some remarks from my side:

Linux 2.1.7, which was released half week ago, now
contains my packet radio drivers for general purpose
sound cards. It works with Soundblaster and WindowsSoundSystem
compatible soundcards. The drivers can do either 1200 baud AFSK
or 9600 baud G3RUH compatible FSK. Since this is Linux,
it comes with sources :-). It should be reasonnably easy
to add FEC for those who want :-)
The driver offers a network driver interface and can thus be
easily used with kernel AX.25, in theory at least. Unfortunately,
kernel AX.25 does not compile in 2.1.7 because of changes
in the user access primitives in the kernel. The drivers may
however be used by user mode ax25 stacks such as WAMPES, which
can now directly access kernel AX.25 stacks.

But to slow down your enthusiasm, here are the top 8 reasons
why a general purpose Soundcard and a PC is _NOT_ a good
signal processing platform:

1. Bad analog section of many soundcards. Frequency response
and/or noise leaves much to be desired.

2. Low resolution or sampling rates are still common. SBPro
can't do 16bits! WSS cards are somehow better...

3. A garden variety of "almost compatible" soundcards makes
life hard. Even Creative's Cards aren't fully backwards
compatible

4. Full duplex cards are still the exception. Creative's
idea of "full duplex" operation is to stick in two soundcards.
This results in ADC and DAC sampling frequencies not being exactly
the same which may cause further difficulties depending on the
application (consider an effect processor. You just cannot output
the data sampled by one soundcard on the other soundcard, you would
have to do complicated sample rate conversions)

5. You cannot usually use the OS driver for the soundcard.
Using it would result in switching times from transmission
to reception and vice versa to be in the seconds range,
which is unacceptable for example for packet radio.
Having to drive the cards yourself exposes you to all the
braindamage of current soundcard architectures

6. It is very nontrivial to attach a time tag to each sample,
i.e. to know or specify exactly when a sample should be output.
This is needed for protocols with a rigid timing scheme, such as
{Am,Pac}Tor. This makes them virtually useless for many HF
protocols.

7. There is not one common sampling frequency every soundcard
can produce, i.e. every brand of soundcard seems to have
a distinct set of sampling frequencies available.

8. Point 7 would not be too bad if one could find out the
actual sampling frequency used. However the actual baud rate
and the rate the soundcard reports to use may differ in
the percent region! This does not matter very much for
audio, but this is bad news for example for PSK.

IMHO it boils down to:

- current soundcards have a braindamaged interface on the
  digital side
- the designers of the analog circuitry seem to be completely
  unaware of digital signal processing.

Of course there are exceptions to this rule. WSS soundcards
seem to be better because the CODEC chip comes from one
of two companies (Analog Devices or Crystal Semiconductors) that
know this field very well.

Tom
hb9jnx/ae4wa

6. Re: [HFSIG:1687] DSP/Modem article

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 04 Nov 1996 09:56:38 -0800
To: hfsig@tapr.org
Message-ID: <199611041755.JAA01919@PEAK.ORG>
Rendered body type: text/plain

>Hi,
>
>Thought I would mention a recent article in Circuit Cellar INK
>(November 1996) that may be of interest.
  ^^^^^^^^^^^^^
Issue #76

http://www.circellar.com/

I have been an avid reader of this very down-to-earth journal
and have gained a lot from it.

--Johan

7. PSAM (IEEE Comm, Nov '96)

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)
Date: Wed, 6 Nov 96 14:58:20 CST
To: hfsig@tapr.org
Message-ID: <9611062058.AA03299@eagle.aud.alcatel.com>
Rendered body type: text/plain

An interesting article in the latest (Nov 96?) issue
of IEEE Communications magazine on 220 Mhz mobile data services.
It discusses multipath fading, and moving vehicles, diversity,
etc.

        One technique discussed is PSAM - Periodic Symbol Assisted
Modulation. The transmitter sends periodically in the frame a known
symbol. Acutally, the periodic symbol is from a set of symbols, so
the periodic symbols do not disturb the modulation stastics too much.

        Even with fast fading of driving a car, the multipath
fading is strictly band-limited. Thus looking at the periodic symbol
gives a way to estimate the channel response, and a way to update
that estimate with time.

        There is an example of a 16-QAM constellation which is
totally useless, but when divided by the channel estimator resolves
into a 16-point constellation quite nicely.

        It would seem that this idea might be extended to HF, since
the multipath on HF is also, I think strictly band limited.   The
periodic symbol allows to compensate well for amplitude variations,
and thus may open up the possibility of the very useful QAM modulation
on HF. 16-QAM is about 4 dB more efficient, in terms of Eb/No, than
16-PSK, but it's tough on HF with the constantly changing amplitude.

        Anyone else seen this article and care to comment on whether it
might be useful on HF?

                                                - Tom

Tom McDermott
mcdermot@aud.alcatel.com

8. Re: [HFSIG:1692] PSAM (IEEE Comm, Nov '96)

From: George Cserenyi <zsolt@direct.ca>
Date: Wed, 6 Nov 1996 15:04:15 -0800 (PST)
To: Tom Mcdermott <mcdermot@rdxsunhost.aud.alcatel.com>
Cc: hfsig@tapr.org
Message-ID: <Pine.LNX.3.91.961106144825.15342A-100000@boxer.direct.ca>
Rendered body type: text/plain

On Wed, 6 Nov 1996, Tom Mcdermott wrote among other things:
>
>   An interesting article in the latest (Nov 96?) issue
> of IEEE Communications magazine on 220 Mhz mobile data services.
> It discusses multipath fading, and moving vehicles, diversity,
> etc.
[...]
>   It would seem that this idea might be extended to HF, since
> the multipath on HF is also, I think strictly band limited.   The
> periodic symbol allows to compensate well for amplitude variations,
> and thus may open up the possibility of the very useful QAM modulation
> on HF. 16-QAM is about 4 dB more efficient, in terms of Eb/No, than
> 16-PSK, but it's tough on HF with the constantly changing amplitude.
[...]
> Tom McDermott
> mcdermot@aud.alcatel.com
>
Haven't got the chance to read that article, but found ur thoughts
on HF multipath interesting.
What practical advantage do you see by using 16-QAM modulation on HF?
What modes would it be most useful for?
What equipment is needed for HF use? Cost? Avaibility?
These are open questions to everyone on the list BTW.

I for one, would like to experiment with it.
Am long on time but short on money since retired from active work.
Is alcatel a swiss company?
73, George
ve7ciz

9. Amplitude calibration

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Wed, 6 Nov 1996 17:40:00 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611070140.AA187840801@polecat.sr.hp.com>
Rendered body type: text/plain

mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott) wrote:

>One technique discussed is PSAM - Periodic Symbol Assisted
>Modulation. The transmitter sends periodically in the frame a known
>symbol. Acutally, the periodic symbol is from a set of symbols, so
>the periodic symbols do not disturb the modulation stastics too much.

>...   The
>periodic symbol allows to compensate well for amplitude variations,
>and thus may open up the possibility of the very useful QAM modulation
>on HF.

It sounds like a very interesting idea. One variation would be the
following: Rather than sending a periodic known symbol, scramble the
input data with a known pseudo-random sequence. That would guarantee
that all possible symbol states occur at approximately equal rates.
Then you could use AGC in the receiver to compensate for fading.

The AGC time constant would have to be long enough to integrate enough
symbols to obtain a good average amplitude. But the time constant would
also have to be short with respect to the fastest fade times. I suspect
that typical values used in HF SSB receivers would be about right, since
they have the same bandwidth ("data rate") and fading constraints.

A similar problem exists with the periodic-symbol approach -- the known
symbol must occur much more often than the fade time, but not so often
that it affects throughput.

A third variation would be to use both techniques together. Scrambling
would allow receiver AGC to keep the input to the demodulator at the
optimum level for best dynamic range, while the periodic known symbol
would allow the demod to dial in a fine amplitude adjustment.

Alan Bloom N1AL

10. Re: [HFSIG:1692] PSAM (IEEE Comm, Nov '96)

From: Charles Brain <chbrain@dircon.co.uk>
Date: Thu, 07 Nov 1996 06:54:57 +0000
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19961107065457.0068b218@popmail.dircon.co.uk>
Rendered body type: text/plain

>   One technique discussed is PSAM - Periodic Symbol Assisted
>Modulation. The transmitter sends periodically in the frame a known
>symbol. Acutally, the periodic symbol is from a set of symbols, so
>the periodic symbols do not disturb the modulation stastics too much.
>
>
Am I missing something here or are we talking about training sequencies,
if so HF serial tone modems use this technique combined with
decision directed equalisation to update their adaptive equalisers.

Charles G4GUO

11. Correction: PSAM (IEEE Comm, Oct '96)

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)
Date: Thu, 7 Nov 96 13:20:01 CST
To: hfsig@tapr.org
Message-ID: <9611071920.AA03465@eagle.aud.alcatel.com>
Rendered body type: text/plain

The 220 Mhz article containing (among other topics) PSAM
(Periodic Symbol-Assisted Modulation) article was in October '96
IEEE Communications magazine. It just arrived and I assumed it
was November, sorry.

- Tom

Tom McDermott
mcdermot@aud.alcatel.com

12. Re: [HFSIG:1695]

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)
Date: Thu, 7 Nov 96 13:23:37 CST
To: hfsig@tapr.org
Message-ID: <9611071923.AA03471@eagle.aud.alcatel.com>
Rendered body type: text/plain

> Am I missing something here or are we talking about training sequencies,
> if so HF serial tone modems use this technique combined with
> decision directed equalisation to update their adaptive equalisers.
>
>
> Charles G4GUO
>

        I guess it depends on the definition of 'training'. I sort
of assume that training applies to time before any data gets sent,
while PSAM applies the symbols interleaved with the data so that
variations in the channel response with time can be tracked out.
In land-line modems, the training time lets the adaptive equalizers
adapt to the channel, but if the channel varies too much and too
quickly after the data has started, the adaption cannot keep up,
and they lose it. Normally not a problem with land-lines, but
perhaps an issue with HF.

- Tom

Tom McDermott
mcdermot@aud.alcatel.com

13. Re: [HFSIG:1694] Amplitude calibration

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)
Date: Thu, 7 Nov 96 13:29:41 CST
To: hfsig@tapr.org

Message-ID: <9611071929.AA03477@eagle.aud.alcatel.com>
Rendered body type: text/plain

> It sounds like a very interesting idea. One variation would be the
> following: Rather than sending a periodic known symbol, scramble the
> input data with a known pseudo-random sequence. That would guarantee
> that all possible symbol states occur at approximately equal rates.
> Then you could use AGC in the receiver to compensate for fading.
>
> The AGC time constant would have to be long enough to integrate enough
> symbols to obtain a good average amplitude. But the time constant would
> also have to be short with respect to the fastest fade times. I suspect
> that typical values used in HF SSB receivers would be about right, since
> they have the same bandwidth ("data rate") and fading constraints.
>
> A similar problem exists with the periodic-symbol approach -- the known
> symbol must occur much more often than the fade time, but not so often
> that it affects throughput.
>
> A third variation would be to use both techniques together. Scrambling
> would allow receiver AGC to keep the input to the demodulator at the
> optimum level for best dynamic range, while the periodic known symbol
> would allow the demod to dial in a fine amplitude adjustment.
>
> Alan Bloom N1AL
>
>

Alan: interesting idea. There are some differences in performance I think. Suitably scrambled data would let you recover phase and amplitude, any QAM modem has to do that. You are restricted to being required to be able to differentiate adjacent constellation points. In the PSAM article, adacent constellation points could not be distuingished. So the PSAM symbols, being known a priori, let one derive the channel response almost exactly even if individual constellation points are not visible.

It was amazing to me that a single 'blob' of a constellation could be resurrected into 16 nice constellation points just by dividing by the computed channel response.

- Tom

Tom McDermott
mcdermot@aud.alcatel.com

14. Re: [HFSIG:1697] Re:

From: Charles Brain <chbrain@dircon.co.uk>
Date: Thu, 07 Nov 1996 20:10:15 +0000
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19961107201015.0067dd78@popmail.dircon.co.uk>
Rendered body type: text/plain

At 13:27 07/11/96 -0600, you wrote:
>
>> Am I missing something here or are we talking about training sequencies,
>> if so HF serial tone modems use this technique combined with
>> decision directed equalisation to update their adaptive equalisers.
>>
>>
>> Charles G4GUO
>>
>
>   I guess it depends on the definition of 'training'. I sort
>of assume that training applies to time before any data gets sent,

Sorry retraining !

You want to ask Harris how they do the equalisation on their serial tone modems.

> Normally not a problem with land-lines, but perhaps an issue with HF.
>
 There's the rub.

>- Tom

73 Charles

15. Amplitude Calibration

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Thu, 7 Nov 1996 14:23:54 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611072223.AA062285434@polecat.sr.hp.com>
Rendered body type: text/plain

mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott) wrote:

>> It sounds like a very interesting idea. One variation would be the
>> following: Rather than sending a periodic known symbol, scramble the
>> input data with a known pseudo-random sequence. That would guarantee
>> that all possible symbol states occur at approximately equal rates.
>> Then you could use AGC in the receiver to compensate for fading.
>
>   Alan: interesting idea. There are some differences in
>performance I think. Suitably scrambled data would let you
>recover phase and amplitude, any QAM modem has to do that.
>You are restricted to being required to be able to differentiate
>adjacent constellation points. In the PSAM article, adacent
>constellation points could not be distuingished. So the PSAM
>symbols, being known a priori, let one derive the channel response
>almost exactly even if individual constellation points are not visible.

So the periodic known symbol is used to recover phase as well as
amplitude information? Many QPSK systems use differential modulation
(where the data bits determine the transitions between symbol states,
rather than the states themselves) to get around the phase ambiguity
problem; PSAM would eliminate that requirement. Scrambling at the
bit rate would not do anything to recover absolute phase information.
Another reason why using PSAM and scrambling together might be a
synergistic solution.

> It was amazing to me that a single 'blob' of a constellation
>could be resurrected into 16 nice constellation points just by
>dividing by the computed channel response.

The clock recovery circuitry in a normal QPSK or QAM detector should
deal with any variations in the phase of the input signal. It seems
like the main advantage of PSAM (or scrambling) is that it can take
out the amplitude variations as well.

Alan Bloom N1AL

16. Re: [HFSIG:1690] Re: DSP/Modem article

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 7 Nov 1996 14:24:34 -0800 (PST)
To: Thomas Sailer <sailer@ife.ee.ethz.ch>
Cc: hfsig@tapr.org
Message-ID: <199611072224.OAA19910@servo.qualcomm.com>
Rendered body type: text/plain

Thomas,

Thanks for your comments. They obviously come from somebody with a lot
of practical experience. Some questions of my own:

>1. Bad analog section of many soundcards. Frequency response
>and/or noise leaves much to be desired.

I agree few cards really make it to CD-quality, but is this really a
problem? I'd expect the radio and the channel to still be the limiting
factors.

>2. Low resolution or sampling rates are still common. SBPro
>can't do 16bits! WSS cards are somehow better...

Again, is this really a problem? Perhaps you can't *totally* ignore
your AF gain setting, but the achievable dynamic range should make it
pretty non-critical, at least with a 16-bit card.

BTW, in Qualcomm CDMA we routinely use 4-bit A/Ds. Analog gain
controls up front bring the signal into the limited dynamic range of
the A/Ds, which is about 24 dB. That roughly matches the processing
gain of the system, and it's plenty for an efficient coded modulation
method.

>3. A garden variety of "almost compatible" soundcards makes
>life hard. Even Creative's Cards aren't fully backwards
>compatible

Yeah, tell me about it. When I wrote my own sound driver for KA9Q NOS,
I decided to support only the SB16. I didn't consider 8-bit support to
be very important since the SB16 was already pretty cheap and seems to
have become the de-facto standard. Everybody seems to be upgrading to
them anyway, to get good quality game audio if nothing else.

>4. Full duplex cards are still the exception. Creative's

True, but this is not a problem with half-duplex radios. Yeah, to do
digital speech you'll need a second card for the speaker/microphone,
though. Too bad it's not possible to put one SB16 channel in receive
and the other in transmit.

>5. You cannot usually use the OS driver for the soundcard.

Why is this? Buffer draining delays? Can the OS buffer size be
configured?

>6. It is very nontrivial to attach a time tag to each sample,

Yeah, I can believe this. One hack might be to tune to WWV to
calibrate both the absolute time and the sample rate. If you have good
low frequency response in the radio you can decode the 100 Hz data
stream and set the PC clock at the same time. But just doing an
autocorrelation on the data sequence ought to bring out the second
ticks and give you a pretty good idea of the sampling rate.

Question: do the soundcards typically maintain their sampling rate
once you know what it is? I note that the clock in my Pentium loses a
very consistent ~7 seconds per day, which tells me that although the
oscillator is off frequency, it's pretty stable.

>7. There is not one common sampling frequency every soundcard
>can produce, i.e. every brand of soundcard seems to have
>a distinct set of sampling frequencies available.

What are these for the SB16? The documentation implies that you can
set an almost continuous range of sample rates.

Phil

17. Re: [HFSIG:1701] Re: DSP/Modem article

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Fri, 08 Nov 1996 10:44:08 +0100
To: hfsig@tapr.org
Cc: karn@qualcomm.com
Message-ID: <328300E8.1F83@ife.ee.ethz.ch>
Rendered body type: text/plain

> >1. Bad analog section of many soundcards. Frequency response
> >and/or noise leaves much to be desired
> I agree few cards really make it to CD-quality, but is this really a
> problem? I'd expect the radio and the channel to still be the limiting
> factors.
Frequency response definitely _is_ a problem. Look at measurements
done by for example the c't magazine. On my laptop, I cannot monitor
G3RUH 9k6 transmissions, due to frequency response problems.
Some soundcards have a built in and not bypassable AGC, which
wreaks havoc (SBPro clones such as ESS).
Furthermore, some soundcards (SB32, presumably also SB16) have
a not bypassable analog "equalizer" filter at the output, which can
induce unknown phase distortions. (When we tried G3RUH FSK, the
best setting of these filter was _not_ what Creative said it was
the 0dB setting)
SB cards have 3 settings for the anti aliasing filter: 3.2kHz,
8.8kHz or off. This is simply not adequate.

> >2. Low resolution or sampling rates are still common. SBPro
> >can't do 16bits! WSS cards are somehow better...>
> Again, is this really a problem? Perhaps you can't *totally* ignore
> your AF gain setting, but the achievable dynamic range should make it
> pretty non-critical, at least with a 16-bit card.
> BTW, in Qualcomm CDMA we routinely use 4-bit A/Ds. Analog gain
> controls up front bring the signal into the limited dynamic range of

> the A/Ds, which is about 24 dB. That roughly matches the processing
> gain of the system, and it's plenty for an efficient coded modulation
> method.
That depends on the system and the signal conditionig prior to the
A/D converter.
A few years ago when I played with my RTTY decoders, the 8bit
soundcard version performed poorly compared to the TI DSK version
(14bit A/D). On HF, if you want to do part of the filtering
digitally, it requires much more than 8 bit resolution.
If you had an analoguely filtered and gain controlled signal,
8bit would be enough, but you presumably would not want to do that,
because good (read with a constant group delay) analog filters
are _extremely_ expensive.
BTW note that the gain controllable amplifier should be controlled
by the demodulator! When I did some experiments with heavily coded
low speed DPSK on HF, week signal performance was often much better
when I turned off the transceivers AGC and adjusted by myself.
AGC circuits of contemporary HF transceivers may be adequate for
SSB, but aren't for digital modes.

> >3. A garden variety of "almost compatible" soundcards makes
> >life hard. Even Creative's Cards aren't fully backwards
> >compatible>
> Yeah, tell me about it. When I wrote my own sound driver for KA9Q NOS,
> I decided to support only the SB16. I didn't consider 8-bit support to
> be very important since the SB16 was already pretty cheap and seems to
> have become the de-facto standard. Everybody seems to be upgrading to
> them anyway, to get good quality game audio if nothing else.
Well, here I do not agree. I even got questions that wondered why
my 9k6 RUH driver does not work with SB2! SBPro is still much more
common than SB16. Anyway I quite dislike SB, see my above comments
on the analog section of the card.
Creative tells us that the only way to do "high speed" (read above
13kSamples/s) AD on SBPro is the highspeed dma command, which
according to creative's SDK does not exist on SB DSP 4 (SB16 and SB32).
SB DSP 4 however has different commands for 8bit DMA, which in
turn do not work on SBPro.
However, when I lately tried the highspeed dma commands on an SB32,
it seemed to work. But this is guesswork...

> >4. Full duplex cards are still the exception. Creative's
>
> True, but this is not a problem with half-duplex radios. Yeah, to do
Oh yes it is. If full duplex would be common, I would have done
some HF modes (Amtor,Pactor) already some time ago. It would make
these HF things much easier.

> >5. You cannot usually use the OS driver for the soundcard.
> Why is this? Buffer draining delays? Can the OS buffer size be
> configured?
As long as you're only receiving (or only transmitting :-))
there is no problem using the OS drivers. But as soon as you'll
have to switch transmission directions in the 10ms region, you're
on your own.
Under Linux, you cannot normally reduce the buffer size to a reasonnably
low value. There is however a method to specify and map the DMA
buffer into the applications address space, but this API is very
complex. And besides, this requires that your process gets scheduled
immediately, which is at least not the case with the 'interactive'
scheduler that is the default. Maybe this works with the realtime
(POSIX.1b) extensions (sched_setscheduler), but I have to admit that
I haven't tried it.
Under Windows, the only way to do rather low delay sound output seems
to be DirectSound, which cannot do sound input.

> >6. It is very nontrivial to attach a time tag to each sample,
>
> Yeah, I can believe this. One hack might be to tune to WWV to
> calibrate both the absolute time and the sample rate. If you have good
> low frequency response in the radio you can decode the 100 Hz data
> stream and set the PC clock at the same time. But just doing an
> autocorrelation on the data sequence ought to bring out the second
> ticks and give you a pretty good idea of the sampling rate.
Is this 100Hz data stream modulated onto WWV? and how?
Of course there are methods to measure the sampling rate,
but how are you going to tell Joe Ham how he should do it?
Too bad if he does not own an HF radio (maybe he only wants to
do VHF/UHF) or happens to live outside the US... My neighbors
at least would definitely not like it if I built an antenna

capable of receiving WWV over here in Europe :-)
(you could say that there's DCF77 here in Europe, but there were
rumours that DCF77 would be shut down in favor of a satellite
TV subcarrier system... ok, one could also use the TV line frequency
as a reference signal)

> Question: do the soundcards typically maintain their sampling rate
> once you know what it is? I note that the clock in my Pentium loses a
> very consistent ~7 seconds per day, which tells me that although the
> oscillator is off frequency, it's pretty stable.
I haven't temperature cycled my soundcards yet, but I besides
this the sampling rate should be stable.

> >7. There is not one common sampling frequency every soundcard
> >can produce, i.e. every brand of soundcard seems to have
> >a distinct set of sampling frequencies available.>
> What are these for the SB16? The documentation implies that you can
> set an almost continuous range of sample rates.
Well yes, according to the documentation, SB16 should be capable of
any integer frequency sampling rate. But I do not own one, so I cannot
tell anything about it. Besides, implementing such a codec having
decent analog performance is not so easy, and since I'm not impressed
by creative's engineering capabilities, I'm rather sceptical.
There are 'continuous rate' codecs by Analog Devices and probably
also by Crystal Semi, and since these firms know their business,
I assume their devices will perform as expected, but with creative...

I'm only talking about the sample I/O capabilities of soundcards,
other systems, such as the FM synthesizer, are IMHO of little
value to implementing communications systems.
IMHO WSS soundcards are usually better than Soundblaster and
compatible, since their main chip is an almost complete
and good soundcard on-a-chip by Analog Devices or Crystal Semi,
so card makers cannot do too much the wrong way.
And these two companies have a tradition and many years of experience
in the quality codec business. Creative's Chipsets on the other hand
leave much to be desired, especially in the audio section.

Tom

18. Re: [HFSIG:1692] PSAM (IEEE Comm, Nov '96)

From: Phil Karn <karn@laptop>
Date: Mon, 4 Nov 96 20:21 PST
To: hfsig@tapr.org
Message-ID: <m0vKd0i-000MG3C@laptop>
Rendered body type: text/plain

Tom,

This "PSAM" stuff sounds an awful lot like training an equalizer on a
periodic known data pattern. And yes, this is a well known and widely
practiced approach to HF data, at least with high performance commercial
and military modems.

Phil

19. Implications of SFDR

From: Johan Forrer <forrerj@peak.org>
Date: Wed, 13 Nov 1996 14:20:56 -0800 (PST)
To: HFSIG@tapr.org
Message-ID: <Pine.SUN.3.91.961113140702.7926A-100000@kira>
Rendered body type: text/plain

Hi all,

Could someone please help me grasp the implications of "spurious free
dynamic range" (SFDR).

1) Dealing with say a band 0 - 20 MHz wide, a pair of signals,
   one at say 9.5 MHz, the other at 10 MHz, where one signal is
   significantly stronger than the other.

2) Dealing with limited number of A/D quatisation levels. What is the
   nature of the noise model?, i.e., band limited, autoregressive, or
   whatever?

3) Does oversampling and decimation have any benifits here, i.e., if
   the A/D convertor could sample a 1 GSPS but have only 3 bits of
   resolution - what is that worth with respect to effective SFDR.

Perhaps you would recognize that these questions relate to working
with DSP at RF rates, i.e., connecting the A/D convertor to the antenna
and hope for the best :-).

Comments, suggestions or discussion would be most helpful.

Thanks much,

--Johan, KC7WW

20. Designing Filters

From: Mike Kerry <100577.1452@CompuServe.COM>
Date: 14 Nov 96 15:09:02 EST
To: "(unknown)" <HFSIG@tapr.org>
Message-ID: <961114200901_100577.1452_GHW117-1@CompuServe.COM>
Rendered body type: text/plain

I am experimenting with designing FIR filters
for HF data modes (aren't we all?) and would be
interested to hear some opinions about the important
features to be considered, e.g:-

- Blackman vs Kaiser vs other windowing functions

- choice of passband width, relating to shift and Baud rate

- perhaps stop-band width is more important than pass-band?

- length of filter, especially considering the effect on ISI

- what else?

Thanks,

Mike Kerry, G4BMK

21. Spurious-Free Dynamic Range

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Thu, 14 Nov 1996 15:37:52 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611142337.AA095324672@polecat.sr.hp.com>
Rendered body type: text/plain

Johan Forrer <forrerj@peak.org> wrote:

>Could someone please help me grasp the implications of "spurious free
>dynamic range" (SFDR).

SFDR is defined as the ratio (usually expressed in dB) between the RMS
value of the input signal and the RMS value of the largest spurious
frequency component. The value depends on the signal waveform, usually
a full-scale sine wave. It gets worse the higher the frequency.
If the signal consists of two sine waves, the test is usually called
"two-tone intermodulation distortion" (two-tone IMD) rather than SFDR.

>1) Dealing with say a band 0 - 20 MHz wide, a pair of signals,
>   one at say 9.5 MHz, the other at 10 MHz, where one signal is
>   significantly stronger than the other.

Data sheet IMD specs assume the two signals are of equal amplitude.
With many analog components, you can assume that third-order spurs
fall 3 dB per dB of the input signal, fifth order spurs fall 5 dB per
dB, etc. But that's not true for A/D converters. Sometimes the spur
actually gets stronger at lower signal levels.

>2) Dealing with limited number of A/D quatisation levels. What is the
>   nature of the noise model?, i.e., band limited, autoregressive, or
>   whatever?

As you probably know, SFDR measures only spurs, not quantization noise.
Signal-to-noise ratio (SNR) in dB is theoretically 6*(bits resolution)
+ 1.76 dB. So a 12-bit A/D theoretically has 73.76 dB SNR. Practical
devices are typically somewhat worse than that.

The noise is white, uniformly spread between 0 Hz and the Nyquist
frequency. If you look at the spectrum using a 1024-point FFT, the
noise will appear 10*log(1024) = 30 dB lower since each frequency

point represents only 1/1024 of the total quantization noise.

>3) Does oversampling and decimation have any benifits here, i.e., if
>   the A/D convertor could sample a 1 GSPS but have only 3 bits of
>   resolution - what is that worth with respect to effective SFDR.

With 3 bits of resolution, the SNR is theoretically 19.76 dB, spread out between 0 and 500 MHz. If you decimated and filtered down to 30 MHz, that would improve the SNR to 19.76 + 10*log(500/30) = 32 dB. Usually, what people do is use delta-sigma A/D converters designed to shape the noise spectrum to be non-white. If most of the noise is pushed up in frequency, then the low-pass decimation filters can get rid of most of it and you end up with much better SNR.

>Perhaps you would recognize that these questions relate to working
>with DSP at RF rates, i.e., connecting the A/D convertor to the antenna
>and hope for the best :-).

Just in the last few months, new A/D converters have been introduced by Analog Devices that make direct sampling of an entire RF band practical. However, they are designed for cellular applications, where the in-band signals are well controlled. That is, unlike the HF amateur bands, all the signals have similar power levels, and there are a limited number of them (<= the number of channels). You still need a tuned RF front end to eliminate out-of-band signals. As I understand it, these new A/Ds are just barely good enough even under these relatively benign conditions. They are used in base-station designs, where price is not much of an issue.

To see what the A/D is up against, consider the US cellular band, which consists of 832 channels between 869 and 894 MHz. I think each A/D only has to deal with one subband ("A" band or "B" band), which consists of 333 channels spaced 30 kHz apart. If each of the incoming 333 signals were exactly the same power level, then the total signal seen by the A/D would have a 10*log(333) = 25 dB peak-to-average ratio. But of course, all the signals are not exactly the same. And the absolute level is not all that well controlled. Unlike analog components, an A/D does not degrade gracefully when the input gets too strong -- once you exceed the input rails, severe distortion results. For both those reasons you need to leave lots of headroom; let's say another 20 dB or so. That puts your signal level down around -45 dB from full scale.

The two-tone IMD spec is not too useful in this application, since there are up to 333 tones to deal with. There are many combinations of third, fifth, seventh, etc. order products from other channels that can produce a spur in a given channel. Two in-phase, equal-amplitude spurs together are 6 dB more powerful than either spur alone. Counting third-order products only, any channel will see 166 spurs from the other 332 channels. The total worst-case spur level would then be up to 20*log(167) = 44 dB stronger than from a single third-order intermod.

So we're talking on the order of 90 dB SFDR for the A/D. And as I recall, cellular equipment manufacturers are looking for numbers even a little better than that. That's right at the state of the art in A/D design.

To cover an amateur HF band, I think even more performance would be called for. For one thing, there can be more than 333 signals on the band at one time. For another, the signal levels vary all over the map. You may wish to listen to a signal just above the noise level (around -130 dBm or so) at the same time as a contest is going on with multiple S9+40 dB (~-30 dBm) signals at the other end of the

band. You would have to adjust the RF gain so that full scale on the A/D is on the order of 0 dBm at the antenna terminals. That implies a SFDR around 130 dB.

And that's for a single band. It would get worse if you wanted to cover 0-30 MHz in one fell swoop.

In addition to spurs, quantization noise would be a problem. Assuming a 16-bit A/D with 90 dB SNR, and a ratio of Nyquist frequency to bandwidth of (500/3), the quantization noise in a 3 kHz channel would be around -112 dBm, or nearly 20 dB louder than the signal you are trying to hear.

Still, that's a lot closer to acceptable performance than existed

a year ago. In years to come, as A/D performance improves and
prices fall, we may yet get all-digital radios with performance
equal to the best analog designs.

Alan Bloom N1AL

22. FIR filters

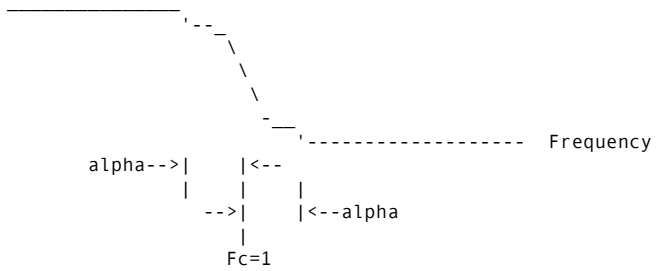
From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Thu, 14 Nov 1996 18:58:53 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611150258.AA233916733@polecat.sr.hp.com>
Rendered body type: text/plain

Mike Kerry <100577.1452@compuserve.com> wrote:

>I am experimenting with designing FIR filters
>for HF data modes (aren't we all?) and would be
>interested to hear some opinions about the important
>features to be considered, e.g:-
>
>- Blackman vs Kaiser vs other windowing functions
>
>- choice of passband width, relating to shift and Baud rate
>
>- perhaps stop-band width is more important than pass-band?
>
>- length of filter, especially considering the effect on ISI
>
>- what else?

One important feature is a Nyquist response. A Nyquist filter is
one where the signal trajectory passes exactly through the proper
symbol location exactly at each decision time. This is true no matter
what values nearby symbols take. i.e. there is no inter-symbol
interference (ISI).

One common Nyquist filter is the "Raised-Cosine." The frequency
response is unity from zero Hz up to (1-alpha) times the cutoff
frequency and zero from (1+alpha) * Fc up to the Nyquist rate.
(Alpha is a selectable parameter that determines how sharply the
filter cuts off.) Between (1-alpha)*Fc and (1+alpha)*Fc, the
frequency response is half of a raised cosine (surprise!):



That's the frequency response. The impulse response in the time domain
is a weighted sinc function. sinc(x) = (sine(PI*x))/(PI*x) Note that
sinc(x) is zero for all non-zero integer values of x. That means that
an impulse (one symbol) has zero effect on adjacent symbols, so there
is no inter-symbol interference.

You need a filter both in the transmitter (to reduce adjacent-channel
interference) and in the receiver (to reject nearby signals). Two
Nyquist filters in series are no longer Nyquist. So what is normally
done is to use a "root-Nyquist" filter such as the root-raised-cosine
filter, whose frequency response is the square root of a raised-cosine.
In this way, the total channel response is Nyquist, so there is no net
inter-symbol interference. Note that the actual transmitted signal has
lots of inter-symbol "interference." It is only after passing through
the matched filter in the receiver that the ISI is magically removed.

By the way, in order to have a Nyquist response, the cut-off frequency
Fc (-3 dB point for root-Nyquist, -6 dB for Nyquist) should be at
exactly 1/2 the symbol rate.

It is possible to use non-Nyquist filters so long as the resulting
inter-symbol interference is kept small enough not to cause a problem.
That's what is done for GMSK. Gaussian Minimum-Shift Keying uses a

(surprise!) Gaussian filter which is non-Nyquist. Commercial systems
that use GMSK include GSM (European digital cellular) and DECT (a
digital cordless phone standard.)

The filter coefficients in an FIR filter are the same as the impulse
response. Since a weighted sinc function theoretically extends from
minus infinity to plus infinity, it is obviously impossible to build a
perfect FIR raised-cosine filter. (The same is true of just about any
other filter, including Gaussian.) One solution is just to truncate
the theoretical response so that it is short enough to fit into your
hardware. If you do that to a Nyquist filter, it remains Nyquist,
since the zero-crossing points remain at the same places. However, it
does affect the frequency response, especially in the stopband, so you
will get additional splatter into adjacent channels. The shorter the
FIR filter length, the worse the interference. It turns out that if
you "window" the response, rather than truncating it, the stopband
rejection is improved.

Windowing means that the coefficients near the edges are gradually
reduced in amplitude so that there is no abrupt transition between full
values and the ones that have been set to zero to fit into the available
filter width. Many different window functions have been invented over
the years. The Hann window (raised cosine in the time domain) is very
simple to implement, but only has 30 dB or so of stopband rejection.
The Hamming window (offset raised cosine) gets down to better than
-40 dB. The Blackman window uses two superimposed raised cosines of
different periods and achieves nearly -60 dB. There is a tradeoff
between stopband and passband performance -- typically the better the
stopband rejection the worse the passband accuracy. The Kaiser window
is parameterised -- by selecting the alpha factor you can optimize the
passband/stopband tradeoff.

A root-Nyquist filter does not have zero crossings at the adjacent
symbol locations. When you truncate or window the impulse response,
inter-symbol interference is created. That's why you don't necessarily
want the window with the best stop-band response -- the poorer passband
accuracy degrdades ISI.

If you want to play around with filters, there are many software programs
available. I like to use Mathcad, even though it is not specialized
for this application. You can choose the filter coefficients using your
favorite window (you have to know the equation for it) or other filter
design method and do an FFT on the coefficients to obtain the frequency
response. Or you can enter the desired frequency response and do an
inverse FFT to obtain the filter coefficients. (Inelegant, but it works!)

You can also calculate the ISI by RMS-summing the time-response values
at all adjacent symbol locations. (If it's a root-Nyquist filter, you
have to convolve it with an "ideal" filter first.)

I think you can get a copy of Mathcad for around $100 or so. It is useful
for many things besides designing filters. It is easy to learn and quite
powerful. I can send a sample FIR filter Mathcad file to anyone interested.

Alan Bloom N1AL

23. Re: [HFSIG:1706] Spurious-Free Dynamic Range

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 14 Nov 1996 19:58:09 -0800
To: hfsig@tapr.org
Message-ID: <199611150354.TAA08791@PEAK.ORG>
Rendered body type: text/plain

Alan,

That was very helpful information - thanks a lot.

I follow the rationale - one further question. I assume IMD originates from
a non-linear process somewhere in the chain, not the quatization process;
that introduces wide-band noise. Am I right? If so, would that imply that if
we had a perfectly linear A/D convertor, and a no clipping situation, that SFDR
ratio would become very large?

The bit about shaping digitization noise for later filtering is quite
interesting. Do you perhaps a have a reference to how that's done?

I recently read a paper on a military radio that uses a one-bit A/D to
sample its 2.5 MHz IF - the high linearity of this A/D attributes to a quoted
SFDR in excess of 105 dB, which is quite usable

As I understand things then, what is needed is a very flexible tunable
bandpass filter ahead of such a A/D.

--Johan

>Johan Forrer <forrerj@peak.org> wrote:
>
>>Could someone please help me grasp the implications of "spurious free
>>dynamic range" (SFDR).
>
>SFDR is defined as the ratio (usually expressed in dB) between the RMS
>value of the input signal and the RMS value of the largest spurious
>frequency component. The value depends on the signal waveform, usually
>a full-scale sine wave. It gets worse the higher the frequency.
>If the signal consists of two sine waves, the test is usually called
>"two-tone intermodulation distortion" (two-tone IMD) rather than SFDR.
>
>>1) Dealing with say a band 0 - 20 MHz wide, a pair of signals,
>>   one at say 9.5 MHz, the other at 10 MHz, where one signal is
>>   significantly stronger than the other.
>
>Data sheet IMD specs assume the two signals are of equal amplitude.
>With many analog components, you can assume that third-order spurs
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>actually gets stronger at lower signal levels.
>
>>2) Dealing with limited number of A/D quatisation levels. What is the
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>+ 1.76 dB. So a 12-bit A/D theoretically has 73.76 dB SNR. Practical
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>frequency. If you look at the spectrum using a 1024-point FFT, the
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>point represents only 1/1024 of the total quantization noise.
>
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>
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>out between 0 and 500 MHz. If you decimated and filtered down to
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>to shape the noise spectrum to be non-white. If most of the noise
>is pushed up in frequency, then the low-pass decimation filters can
>get rid of most of it and you end up with much better SNR.
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>>Perhaps you would recognize that these questions relate to working
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>>and hope for the best :-).
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>by Analog Devices that make direct sampling of an entire RF band
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>HF amateur bands, all the signals have similar power levels, and there
>are a limited number of them (<= the number of channels). You still
>need a tuned RF front end to eliminate out-of-band signals. As I
>understand it, these new A/Ds are just barely good enough even under
>these relatively benign conditions. They are used in base-station
>designs, where price is not much of an issue.
>
>To see what the A/D is up against, consider the US cellular band,
>which consists of 832 channels between 869 and 894 MHz. I think
>each A/D only has to deal with one subband ("A" band or "B" band),
>which consists of 333 channels spaced 30 kHz apart. If each of
>the incoming 333 signals were exactly the same power level, then
>the total signal seen by the A/D would have a 10*log(333) = 25 dB
>peak-to-average ratio. But of course, all the signals are not
>exactly the same. And the absolute level is not all that well

>controlled. Unlike analog components, an A/D does not degrade
>gracefully when the input gets too strong -- once you exceed the
>input rails, severe distortion results. For both those reasons
>you need to leave lots of headroom; let's say another 20 dB or so.
>That puts your signal level down around -45 dB from full scale.
>
>The two-tone IMD spec is not too useful in this application, since
>there are up to 333 tones to deal with. There are many combinations
>of third, fifth, seventh, etc. order products from other channels
>that can produce a spur in a given channel. Two in-phase,
>equal-amplitude spurs together are 6 dB more powerful than either
>spur alone. Counting third-order products only, any channel will
>see 166 spurs from the other 332 channels. The total worst-case
>spur level would then be up to 20*log(167) = 44 dB stronger than
>from a single third-order intermod.
>
>So we're talking on the order of 90 dB SFDR for the A/D. And as I
>recall, cellular equipment manufacturers are looking for numbers
>even a little better than that. That's right at the state of the
>art in A/D design.
>
>To cover an amateur HF band, I think even more performance would be
>called for. For one thing, there can be more than 333 signals on
>the band at one time. For another, the signal levels vary all over
>the map. You may wish to listen to a signal just above the noise
>level (around -130 dBm or so) at the same time as a contest is going
>on with multiple S9+40 dB (~-30 dBm) signals at the other end of the
>band. You would have to adjust the RF gain so that full scale on
>the A/D is on the order of 0 dBm at the antenna terminals. That
>implies a SFDR around 130 dB.
>
>And that's for a single band. It would get worse if you wanted to
>cover 0-30 MHz in one fell swoop.
>
>In addition to spurs, quantization noise would be a problem.
>Assuming a 16-bit A/D with 90 dB SNR, and a ratio of Nyquist
>frequency to bandwidth of (500/3), the quantization noise in a
>3 kHz channel would be around -112 dBm, or nearly 20 dB louder
>than the signal you are trying to hear.
>
>Still, that's a lot closer to acceptable performance than existed
>a year ago. In years to come, as A/D performance improves and
>prices fall, we may yet get all-digital radios with performance
>equal to the best analog designs.
>
>Alan Bloom N1AL
>
>

24. [HFSIG:1707] FIR filters, demodulation

From: Mike Kerry <100577.1452@CompuServe.COM>
Date: 15 Nov 96 15:43:45 EST
To: "(unknown)" <HFSIG@tapr.org>
Message-ID: <961115204345_100577.1452_GHW126-1@CompuServe.COM>
Rendered body type: text/plain

Thanks Alan (N1AL) for the detailed reply. One thing however,
you said:-

> You need a filter both in the transmitter (to reduce adjacent-channel
> interference) and in the receiver (to reject nearby signals).

Does this imply there is no point in trying to use Nyquist receive
filters when decoding sundry Amtor Pactor RTTY signals which may
have been generated by non-Nyquist-compatible means? If so,
then what!?

Have you the time to elaborate on some phrases in your reply?
The first is "a Nyquist filter is one where the signal trajectory
passes exactly thru the proper symbol location exactly at decision
time". What is the signal trajectory? And proper symbol location?

Also: "for a Nyquist response the cut-off frequency should be at 1/2
the symbol rate".

So for 100 Bd Amtor, Fc is 50 Hz ??? I don't understand. Sorry!

The second half of your message on windowing functions and Mathcad
was helpful, tnx.

If I may ask another broad question, may I elicit views on demodulation techniques for RTTY Amtor Pactor? e.g. FM versus AM demodulators, or what else?

Thanks,

Mike Kerry, G4BMK

25. [DSP:36] GCC56K

From: Mike Kerry <100577.1452@CompuServe.COM>
Date: 15 Nov 96 15:43:48 EST
To: "(unknown)" <HFSIG@tapr.org>
Message-ID: <961115204348_100577.1452_GHW126-2@CompuServe.COM>
Rendered body type: text/plain

Tim said:

> You can get the MS-DOS executable from

> ftp://www.mot.com/SPS/DSP/helpline/cc/dist.wpc.cc56.tar.z

Tim, I tried, but was told the page does not exist.
I searched the Motorola DSP index for GCC56K but it found nothing.

Mike, G4BMK

26. SFDR

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Fri, 15 Nov 1996 13:36:17 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611152136.AA053593777@polecat.sr.hp.com>
Rendered body type: text/plain

forrerj@peak.org (Johan Forrer) wrote:

>I assume IMD originates from
>a non-linear process somewhere in the chain, not the quatization process;
>that introduces wide-band noise.

Right, the quantization is generally modeled as white noise, although it can depend on the input signal. For example, if the input is a sine wave a little more than one LSB in amplitude, then the quantized output will appear as a square wave, which looks like distortion, rather than noise. Intuitively, if the input signal is "large" and "random" in some sense, then the quantization error will be noise-like.

>Am I right? If so, would that imply that if
>we had a perfectly linear A/D convertor, and a no clipping situation, that
>SFDR ratio would become very large?

That's right. Many of the nonlinearities are dynamic effects that don't show up with low-frequency or DC signals. And of course, not only does the A/D converter have to be linear, but so does any other analog circuitry between the antenna and A/D input.

>The bit about shaping digitization noise for later filtering is quite
>interesting. Do you perhaps a have a reference to how that's done?

When I get home tonight I'll check to see if the topic is covered in a couple DSP books I have at home.

>I recently read a paper on a military radio that uses a one-bit A/D to
>sample its 2.5 MHz IF - the high linearity of this A/D attributes to a quoted
>SFDR in excess of 105 dB, which is quite usable

One-bit delta-sigma A/D converters tend not to have very good absolute accuracy, but they can have very good linearity (low distortion). That's why they are often used in audio applications -- distortion is much more important than gain accuracy. The same would apply to RF applications. However, the analog sampling rate must be much higher than the digital output samples (and much higher than the RF bandwidth), which of course is much harder to do at RF than at audio frequencies.

>As I understand things then, what is needed is a very flexible tunable
>bandpass filter ahead of such a A/D.

That would help a lot. You would then essentially have a TRF (Tuned

Radio Frequency) receiver like they used in the 1920's, except that the crystal detector is replaced by a DSP! The other option is to do have a conventional superhet receiver front end with a relatively wide-bandwidth IF crystal filter and a digital DSP-based IF. The DSP could implement variable-bandwidth bandpass filters, most of the AGC function, noise blanker, notch filter, and demodulator. Compared to the all-digital radio, the A/D and DSP's jobs are much easier, because of the narrower bandwidth and reduced dynamic range requirements (assuming there is some analog gain control in front of the A/D.)

By the way, the sampling rate must be > 2x the bandwidth, not 2x the frequency. For example, if the IF frequency is 2.5 MHz, and the bandwidth is 100 kHz, you can theoretically use sample rates as low as 200 kHz. (Although in practice it has to be somewhat higher than that.) With an I/Q demodulator, sample rate must be > 1x bandwidth.

Al N1AL

27. Re: [HFSIG:1710] [DSP:36] GCC56K

From: Tim Baggett <tim@acca.nmsu.edu>
Date: Fri, 15 Nov 1996 19:16:29 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.95.961115191349.26474A-100000@acca>
Rendered body type: text/plain

On Fri, 15 Nov 1996, Mike Kerry wrote:

> Tim said:
>
> > You can get the MS-DOS executable from
>
> > ftp://www.mot.com/SPS/DSP/helpline/cc/dist.wpc.cc56.tar.z
>
> Tim, I tried, but was told the page does not exist.
> I searched the Motorola DSP index for GCC56K but it
> found nothing.

How in the world did this get over to the HFSIG list??!! I thought it was on the DSP list were I get it at work!

Anyhow, leave the filename off. I guess netscrape doesn't like a URL that goes for the specific file. Chop it off after the /cc and you will get the directory.

BTW, the manual gcc56kman.zip or something like that is in a stupid FrameMaker format. I have one of the docs guys converting it over to PDF format. They tend to think everyone in the world has Frame Maker. I never heard of it until I started work there.

73
Tim

28. More on filters

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Fri, 15 Nov 1996 18:19:14 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611160219.AA208680754@polecat.sr.hp.com>
Rendered body type: text/plain

Mike Kerry <100577.1452@compuserve.com> wrote:

>you said:-
>
>> You need a filter both in the transmitter (to reduce adjacent-channel
>> interference) and in the receiver (to reject nearby signals).
>
>Does this imply there is no point in trying to use Nyquist receive
>filters when decoding sundry Amtor Pactor RTTY signals which may
>have been generated by non-Nyquist-compatible means? If so,
>then what!?

Ideally, the filters in the receiver and transmitter should be matched so that the net response is Nyquist. You can always find a receive filter that accomplishes that by adding an equalizing filter whose response is the reciprocal of transmit filter response. (Theoretically, anyway.) But of course you have to know the transfer function of the transmit filter.

Fortunately the filters don't have to be perfect. People got along
fine with analog filters (inherently non-Nyquist) for years. Of course,
they had to make do with wider bandwidths than necessary to keep the
inter-symbol interference within reason.

>Have you the time to elaborate on some phrases in your reply?
>The first is "a Nyquist filter is one where the signal trajectory
>passes exactly thru the proper symbol location exactly at decision
>time". What is the signal trajectory? And proper symbol location?

By "signal trajectory" I just mean a plot of signal versus time. At
the receiver detector (after filtering), the instantaneous value of the
signal is measured at the decision time, once per symbol. The possible
symbol locations depend on the modulation type. For FSK or BPSK, there
are only two possibilities. For 256 QAM, there are 16 possible symbol
locations for the I channel and 16 for the Q channel, giving a total of
16*16 = 256 possible symbol locations on the I/Q plane.

The job of the detector is to read the signal value at the decision
time and decide which of the possible symbol locations the signal is
closest to. For FSK, if the symbol locations are +1 and -1, then any
signal value above 0 is considered a +1 and anything below is a -1.

>Also: "for a Nyquist response the cut-off frequency should be at 1/2
>the symbol rate".
>
>So for 100 Bd Amtor, Fc is 50 Hz ??? I don't understand. Sorry!

Yes. Theoretically, a signal doesn't need to have any energy above
1/2 the symbol rate. Consider a 1-bit-per-symbol modulation like FSK.
The fastest you can make that signal change is to send it a continuous
series of alternating one's and zero's. So if the baud rate is 100
symbols/sec, the unfiltered signal would be a 50 Hz square wave.
* (See disclaimer below) If you filtered it with a brick-wall filter
with a cutoff at 50 Hz (i.e. a Nyquist filter with an alpha = 0), the
square wave would be turned into a sine wave, and no information would
be lost. Any other combination of 1's and 0's can only have even lower
frequency components. For example 110011001100... has a repetition rate
of Fs/4.

Of course, a practical filter can't have a brick-wall frequency
response. But to be Nyquist (no inter-symbol interference), it must
be half-scale (-6 dB) at 1/2 the symbol rate and be anti-symmetrical
about that point on the frequency axis. A raised-cosine filter is one
that fits that description.

>If I may ask another broad question, may I elicit views on demodulation
>techniques for RTTY Amtor Pactor? e.g. FM versus AM demodulators, or
>what else?

AM demodulators are supposed to work better than FM discriminators.
Years ago, I used a Frederick Electronics demodulator quite a bit on
the HF bands. It could be switched between FM and AM modes. I could
never see much difference, as a practical matter. As I recall, that
demod had circuitry to dynamically adjust the decision level of the
"slicer" (mark/space comparator), to account for selective fading
(one tone stronger than the other) and interference affecting one
frequency more than the other. The box did indeed seem to work better
than standard designs.

Al N1AL

29. unsubscribe

From: beppe@ns1.dinamica.it
Date: Sat, 16 Nov 1996 12:03:27 +0100
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19961116113718.096fa3ea@pt.dinamica.it>
Rendered body type: text/plain

==========
Giuseppe Selvatici beppe@pt.dinamica.it
==========

30. Re: [HFSIG:1709] FIR filters, demodulation

From: forrerj@peak.org (Johan Forrer)
Date: Sat, 16 Nov 1996 19:51:57 -0800
To: hfsig@tapr.org
Message-ID: <199611170348.TAA02151@PEAK.ORG>

Rendered body type: text/plain

Mike,

My 2 cents on the subject;

>
>If I may ask another broad question, may I elicit views on demodulation
>techniques for RTTY Amtor Pactor? e.g. FM versus AM demodulators, or
>what else?
>
>Thanks,
>
>Mike Kerry, G4BMK
>
>

My experience has been that a MAP receiver exhibits exceptional robustness
for HF operation. Such a MAximum Postiori receiver may be considered a
filter bank and a selector switch that selects the filter output with
largest magnitude. There are some elegant mathematics to support this, which
can be extended to produce matched filter theory.

Good adjacent channel interference is of prime importance.

With synchronous data systems, such as AMTOR and Pactor, the demodulator and
detector operation can be controlled as a byproduct from the data decoding
process. In this case a software PLL can guide the demodulator/detector to
clean up data decisions. The performance is not as good as a real coherent
demodulator, but can approach that.

A novel idea is to make use of soft-decision decoding, i.e., Viterbi
decoding to aid in bit synchronisation and data extraction. An example where
this have been done successfully is shown in the work of Dave Mills, W3HCF
where he uses such a scheme to decode RTTY and FEC AMTOR. This by itself is
worth a couple of dB and also buys further robustness.

Yes, it is possible to decode FSK using phase information, i.e., multiply
the signal by a delayed version of itself. The FSK demodulators that Pawel,
SP9VRC wrote for the EVM does this, so does the DSP93 modems of Frank
Perkins. It works reasonably well, howeber, I am of the opinion that this
technique does not show the same robustness as the filter-type approach
described above. Possibly it is a tradeoff between phase stability/ambiuity
against spectral power measurement. I suspect the latter being a more
statistically robust measure.

Hope this helps.

--Johan, KC7WW

31. [HFSIG:1713] Filters and Demods

From: Mike Kerry <100577.1452@CompuServe.COM>
Date: 17 Nov 96 18:55:06 EST
To: "(unknown)" <HFSIG@tapr.org>
Message-ID: <961117235505_100577.1452_GHW48-1@CompuServe.COM>
Rendered body type: text/plain

Thanks to Alan and Johan for their contributions.

I have been experimenting with an FM demodulator, and
also a matched filter approach, taking the difference
in magnitude between mark and space filters (not sure if
that constitutes an AM decoder or what. Someone is bound
to enlighten me).

The FM system is currently giving better results on higher
baud rates (e.g. 200 Bd Pactor) and the matched filter
approach is better on RTTY. (These both being compared to a
BARTG Multyterm matched filter op-amp terminal unit).

I like the idea of running multiple filters in parallel
and choosing the set that gives maximum eye-height or
difference - or of adjusting filter coefficients on the fly
to maximise. With modes like Pactor and AX25 especially,
with frame checking being done in the DSP too, one can run
multiple techniques in parallel and just see which gives you a
valid CRC for the frame. This would be nice for monitoring
Pactor, as you can effectively be continuously running
optimally at both 100 and 200 Bd.

With non self-checking modes like RTTY, is there a danger of
other factors like QRM affecting eye-height and confusing
the filter choice, or is eye-height or matched filter
difference always going to be a good guide?

Can someone tell me please how to compute the optimum receive
filter pass-band width for a given Baud rate and Shift, to
maximise legitimate energy capture. And are you talking about
3 dB or 6 dB points.

Some background information:
My DSP platform at present is a 20 MIPS TMS320C51 processor
with TLC32044 AIC, 64K full speed SRAM and 64K Flash. I have the
TI Coff Assembler, and 'C' Compiler (rather bugged!). Main uses
to date have been for 1200 Baud FSK and 2400 Baud QPSK used
over trunked radio networks. Now looking at the HF modes.
I am awaiting delivery of my DSP56002EVM.

73,

Mike Kerry, G4BMK

32. TU-170

From: k5cnf1@juno.com (Richard B Morrow)
Date: Sun, 17 Nov 1996 08:15:44 PST
To: HFSIG@tapr.org
Message-ID: <19961117.082836.8279.0.k5cnf1@juno.com>
Rendered body type: text/plain

anyone out there have any idea as to how I could put an ancient Flesher
TU-170 to use with my newer computers. The only program I have is for
the ancient Radio Shack Model 1 which is not an option to use. I have
the docs on the tu, which helps.

Richard, K5CNF

33. [HFSIG:1717] TU-170

From: Mike Kerry <100577.1452@CompuServe.COM>
Date: 18 Nov 96 05:50:39 EST
To: "(unknown)" <HFSIG@tapr.org>
Message-ID: <961118105038_100577.1452_GHW47-1@CompuServe.COM>
Rendered body type: text/plain

Richard, K5CNF wrote:

>anyone out there have any idea as to how I could put an ancient Flesher
>TU-170 to use with my newer computers. The only program I have is for
>the ancient Radio Shack Model 1 which is not an option to use. I have
>the docs on the tu, which helps.

Richard, try contacting Steve, AC4IW, at Spheretron. He can supply
the BMK-Multy software which does RTTY, Amtor and Pactor etc. on a PC
and I'm pretty sure the TU-170 is amongst TU's that have been interfaced.

Spheretron 25 Eastwood Road, P.O.Box 5964, Asheville NC 28813
Tel: (704) 274 4646

Mike, G4BMK

34. Re: [HFSIG:1717] TU-170

From: w5zit@juno.com (James H Brown)
Date: Mon, 18 Nov 1996 11:03:01 EST
To: hfsig@tapr.org
Message-ID: <19961118.110303.7879.1.w5zit@juno.com>
Rendered body type: text/plain

Richard - the TU-170 was designed for 45 Baud RTTY - but has the makings
of a front end for the faster Baud rates if you do a little tinkering
with it.

Take a look at the filtering after the mark/space detector and see if the
output transitions are restricted to a slower rate. You may have to
decrease the value of one or more filter caps to get the higher data
rates.

There are several packages that run RTTY from a PC that would require no

mod to the TU-170 at all - except for perhaps an adapter to get the
RS-232 levels conditions for your PC. I don't recall if the I/O of the
TU-170 is RS-232 compatable or just TTL - but you probably have all the
specs there in your docs.

I have used an Amtor freebie program on HF with some success, and the
TU-170 should work well on Amtor with the Baud filter modified to pass
the 100 Baud data. On the other hand - if the Flesher specs say it was
designed to pass 100 Baud RTTY signals - then no modification would be
required for Amtor.

There is a program that can be purchased to run Pactor to an external
modem also - which would require modifying the TU-170 Baud fliter to pass
the 200 Baud Pactor signals. I do not recall the name of the program -
but I have worked several hams using it.

Look on the QRZ CD Rom for programs that run on the PC to run various
modes. You may have some ham buddies who have that CD who can help.

If you have trouble locating any of the specific programs - let me know
and I will see what I can dig up.

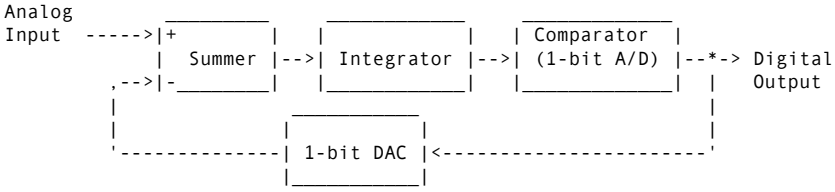
73 - Jim - w5zit@juno.com

35. Sigma Delta

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Mon, 18 Nov 1996 14:56:44 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611182256.AA067387805@polecat.sr.hp.com>
Rendered body type: text/plain

I promised to try to find a reference explaining Sigma-Delta A/D converters.
None of my DSP textbooks seem to cover the topic. I did find an app note
from Harris Semiconductor (AN9504, May 1995) that does a pretty good job.
I can copy it if anyone is interested.

Basically a sigma-delta converter is formed thus:



The digital output is a 1-bit signal at a high bit rate. It then goes to
a low-pass filter and decimator to produce an N-bit output at a lower rate.

Let's say the 1-bit DAC puts out +1V for a "1" and -1V for a "0".
(Actually you'd probably just use a CMOS gate that puts out +5V and 0V,
if you don't mind having a unipolar analog input.) If the analog input
is 0V, then the feedback loop forces the digital output to be half 1's
and half 0's so that the integrator input sees half +1V and half -1V.
The output of the integrator is then a small triangle wave centered
about zero volts. If the analog input is +0.8V (90% full scale), then
the digital output is 90% 1's and 10% 0's.

Because there is an integrator (-6 dB per octave frequency response) in
the feedback path, the quantization noise at the digital output has a
+6 dB/octave rising characteristic, dropping to minus infinity (zero
error) at DC.

If you are using a large oversample ratio (high sampling rate), then
the digital filter in the decimator can filter off most of that high-
frequency quantization noise. Even though a 1-bit A/D theoretically
only has 6 + 1.76 = 7.76 dB S/N ratio, after filtering it can be
equivalent to, for example, a 16-bit A/D. The digital filter has a
1-bit input and an N-bit output.

By using a second-order converter with two integrators, you get a
12 dB/octave frequency response. Third-order gives 18 dB/octave, etc.

The advantage of the sigma-delta A/D is that the linearity is not
affected by nonlinearities in the internal ciruitry of a conventional
multi-bit A/D. Also, most of the circuitry is digital, rather than analog,
which makes it easier to design the whole thing into a monolithic IC.

Alan Bloom N1AL

36. Re: [HFSIG:1718] TU-170

From: k5cnf1@juno.com (Richard B Morrow)
Date: Mon, 18 Nov 1996 06:38:42 PST
To: hfsig@tapr.org
Message-ID: <19961118.065406.8279.1.k5cnf1@juno.com>
Rendered body type: text/plain

OK MIKE Ha! had the caps lock on, you are the second person to recommend
that program..

Thanks a lot.

Richard, K5CNF

37. (no subject)

From: (unknown sender)
Date: (date unavailable)
To: hfsig@tapr.org
Rendered body type: text/plain

Date: Mon, 18 Nov 1996 06:43:56 PST
Subject: Re: TU-170
Message-ID: <19961118.065406.8279.2.k5cnf1@juno.com>
References: <19961118.110303.7879.1.w5zit@juno.com>
X-Mailer: Juno 1.00
X-Juno-Line-Breaks: 2-6
From: k5cnf1@juno.com (Richard B Morrow)

Ok, Jim, I know some one who has that CD.ROM. I have the capabilities of
being able to look at the signals in the TU, several scopes,so I will
check that stuff out

Thanks loads

Richard, K5CNF

38. (no subject)

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 18 Nov 1996 18:47:23 -0800 (PST)
Message-ID: <199611190247.SAA09847@servo.qualcomm.com>
Rendered body type: text/plain

To: hfsig@tapr.org
In-reply-to: <199611170348.TAA02151@PEAK.ORG> (forrerj@peak.org)
Subject: Re: [HFSIG:1715] Re: FIR filters, demodulation

I would like to add to Johan's and Alan's excellent comments by giving
a little fundamental theory, as it's essential in understanding their
remarks. Otherwise, all that complexity about Nyquist and raised
cosine responses seems baffling.

Two requirements must be satisfied simultaneously to maximize the
performance of a digital link.

First, the receiver filter must be "matched" to the transmitted signal
spectrum. That is, the receiver filter amplitude response must match
the amplitude of the transmitted signal spectrum. Additionally, if
there is group delay distortion (frequency-dependent delay) in the
transmitted signal, as can happen with sloppy analog filter design in
the modulator, then the matched filter should have the inverse delay
curve to compensate for the transmitter filter.

Second, the combination of the transmitter filter and the receiver
filter should have a Nyquist response, i.e., the group delay is flat
and the frequency response is anti-symmetrical around the frequency
corresponding to one-half the data rate.

The first requirement maximizes the SNR performance against noise and
interference while the second requirement minimizes self-interference,
i.e., intersymbol interference, at the important moment in the center
of each bit when you sample it. Note that the ISI is *not* zero on
either side of the correct sampling instant, which is why it's
important to have accurate clock recovery. In an eye diagram, this ISI
shows up as "jitter" in the eye on either side of the center.

The "square-root-of-raised-cosine" approach comes about as follows:

A raised cosine channel response satisfies requirement #2
above. Though it is not the only response to do so (anything with the
proper symmetry around 1/2 the data rate will do), it is the easiest
to implement and to model. In particular, a brick-wall filter that
cuts off precisely at 1/2 of the data rate will also do, but
brick-wall filters are notoriously impossible to realize in practice.

Second, if you take the square root of the raised cosine response and
implement it in both the transmitter and the receiver, then the
receiver's response will be matched to the transmitter's spectrum,
meeting requirement #1 above. Furthermore, the combination of the two
square-root responses, when multiplied together, gives the original
raised-cosine response, which continues to meet requirement #2.

This assumes that data bits are sent into the transmitter filter as
impulses of infinite amplitude and infintesmal duration. As such
impulses have a uniformly flat frequency spectrum, the spectrum of the
resulting signal will be that of a square root of raised cosine -- to
which the receiver filter is matched, because it also has a square
root of raised cosine response.

In a real transmitter that uses DSP, these impulses can be closely
approximated by single-sample impulses. E.g., if the sampling rate is
8x the bit rate (a common choice) then the data stream fed into the
transmitter filter might look like this:

-1 0 0 0 0 0 0 0 +1 0 0 0 0 0 0 0 -1 ...

where the data is bipolar (1 remains 1 and 0 becomes -1).

As an alternative (often done in hardware) the data signal is held for
the duration of each bit, e.g.,

-1 -1 -1 -1 -1 -1 -1 -1 +1 +1 +1 +1 +1 +1 +1 +1 -1 ...

In this case, the spectrum of the signal is *not* flat -- it has a sin(x)/x
shape, so you need a x/sin(x) correction factor in the transmitter filter.
The filter design programs often give this to you as an option, e.g.,
as a x/sin(x) * sqrt(raised cosine) response. My own preference so far
for DSP filters is to use the data impulse approach so the transmitter
and receiver filters can be identical.

Now all this assumes you have control over both the receiver and
transmitter. But if you're designing a receiver for an arbitrary RTTY
transmitter, you're going to have to be more flexible. The transmitter
might not have been designed to have a square-root-of-raised-cosine
response. This means you'll generally have to use a suboptimum filter
(from a matched filter standpoint) at the receiver in order to
eliminate ISI on the desired signal, or vice versa. The demod *will*
work, but you'd have worse SNR performance than you'd like.

Alternatively, you could figure out what the transmitter filter looks
like, then make a matched filter at the receiver. This would leave
ISI, which you could take out with a Viterbi decoder (assuming the
duration of the ISI is not too great, which would otherwise entail a
Viterbi decoder with an impractical number of states).

Still another possibility is to build an equalizer to deal with the
nonideal transmitter filter, but this gets quite hairy and advanced.

This is why I personally find it more interesting to design new links
from scratch where I can control both ends of the link. Not only can I
make the frequency responses optimum, but I can add FEC and control
other aspects of the link protocol.

Phil

39. Clover and the P-38

From: "Ronald H. Evans" <ronald_evans@pipeline.com>
Date: Tue, 19 Nov 1996 00:35:03 -0500
To: hfsig@tapr.org
Message-ID: <2.2.32.19961119053503.007033b0@pop.pipeline.com>
Rendered body type: text/plain

Is this the place to discuss Clover, Pactor II, software to use with my P-38
and related topics? I am especially interested in battery powered portable
operation of these modes. Tnx, Ron, K4KTB

40. RE: [HFSIG:1724] Clover and the P-38

From: Peter Schulze <dts@dtsdata.intnet.bj>
Date: Tue, 19 Nov 1996 08:00:32 +-100
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BBD5EF.C0226440@newserver.dtsdata>
Rendered body type: text/plain

The IDRA reflector discussed these issues, check

http://dtsdata.intnet.bj/idra/newsub.htm

for details. Previous articles on that subject are archived at:

http://dtsdata.intnet.bj/idralog/subject.htm

73
Peter TY1PS

41. RTTY

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Tue, 19 Nov 1996 13:41:10 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611192141.AA095919670@polecat.sr.hp.com>
Rendered body type: text/plain

Phil Karn <karn@qualcomm.com> wrote:

> As an alternative (often done in hardware) the data signal is held for
>the duration of each bit, e.g.,
>
>-1 -1 -1 -1 -1 -1 -1 -1 +1 +1 +1 +1 +1 +1 +1 +1 -1 ...
>
>In this case, the spectrum of the signal is *not* flat -- it has a sin(x)/x
>shape, so you need a x/sin(x) correction factor in the transmitter filter.
...
>Now all this assumes you have control over both the receiver and
>transmitter. But if you're designing a receiver for an arbitrary RTTY
>transmitter, you're going to have to be more flexible. The transmitter
>might not have been designed to have a square-root-of-raised-cosine
>response. This means you'll generally have to use a suboptimum filter
>(from a matched filter standpoint) at the receiver in order to
>eliminate ISI on the desired signal, or vice versa. The demod *will*
>work, but you'd have worse SNR performance than you'd like.
...
>Still another possibility is to build an equalizer to deal with the
>nonideal transmitter filter, but this gets quite hairy and advanced.

Most traditional RTTY transmitters get around the ISI problem simply by
using a filter cutoff frequency much higher than needed. This results
in a wider transmitted bandwidth, but at 45 or 100 baud, who cares?
Most receivers don't have a narrow enough crystal filter to make use of
signals separated by 100 Hz anyway (and would have horrible group delay
variation if they did.)

In designing the receiver, you wouldn't go far wrong to assume that
there is no transmit filter and that each mark or space tone is held
for the entire duration of the symbol as Phil illustrated above. The
receiver filter can include the sin(x)/x correction and a Nyquist filter
convolved together into one set of filter coefficients to get a net
almost-Nyquist response.

To generate the FIR filter coefficients, the easiest way would be to
multiply the raised-cosine frequency response with PI*x/sin(PI*x) and
do an inverse FFT on the result to get the impulse response, which is
the FIR filter coeficients. Then apply your favorite window function
to truncate to the length of your filter.

Another refinement would be to cancel out the amplitude and group delay
variation in the receiver crystal filter. You would like to use as
narrow a filter as possible to eliminate interference, but narrow
analog filters add ISI. To measure the filter response you could
generate a narrow RF pulse (use the Woodpecker as a test signal? :=)
and use the demodulator's A/D to measure the impulse response. The
reciprocal of that would then be convolved with the receiver filter
coefficients as determined above. When is some manufacturer going to
design a demod with this feature built in?

Alan Bloom N1AL

42. Re: [HFSIG:1711] SFDR

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 19 Nov 1996 21:09:39 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611200509.VAA02017@servo.qualcomm.com>
Rendered body type: text/plain

>By the way, the sampling rate must be > 2x the bandwidth, not 2x the
>frequency. For example, if the IF frequency is 2.5 MHz, and the bandwidth

One comment here -- although the A/D converter can take its time when
the IF bandwidth is much less than the frequency, the sample-and-hold
used ahead of the A/D converter must still have a very short sample
time compared to the period of the RF waveform. As I understand it,
the sample-and-hold circuits are one of the limiting factors on the
performance of a digital radio that uses direct IF sampling.

Phil

43. KISS explanations

From: Marijan Miletic <s56a@s55tcp.ampr.org>
Date: Wed, 20 Nov 96 08:56:31 EST
To: hfsig@tapr.org
Message-ID: <14942@s55tcp.ampr.org>
Rendered body type: text/plain

Alan's, N1AL explanation of sigma-delta explanation reminded me of even simler
circuit diagram using comparator, D-type flip-flop and RC feedback in some old
Motorola book on telecommunicattion IC's. Many DSP topics are explained in
straightforward fashion in TI application books while the others involve a lot
of math to fill the pages. However, when the going gets tough, one has to
revert to abstract math :-)
73 de Mario, S56A, N1YU.

44. S/H

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Wed, 20 Nov 1996 10:38:17 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611201838.AA089115098@polecat.sr.hp.com>
Rendered body type: text/plain

Phil Karn <karn@qualcomm.com> wrote:

>>By the way, the sampling rate must be > 2x the bandwidth, not 2x the
>>frequency.
>
>One comment here -- although the A/D converter can take its time when
>the IF bandwidth is much less than the frequency, the sample-and-hold
>used ahead of the A/D converter must still have a very short sample
>time compared to the period of the RF waveform. As I understand it,
>the sample-and-hold circuits are one of the limiting factors on the
>performance of a digital radio that uses direct IF sampling.

Good point. That's probably why a lot of radios with digital IF's
go to the trouble of heterodyning down to a low IF frequency before
digitizing. Designing a high-speed sample-and-hold is not trivial.

But the S/H is the only part that has to run at 2x the IF frequency.
The A/D and the DSP can run at the lower rate (2x bandwidth).

Al N1AL

45. RE: [HFSIG:1729] S/H

From: Rick Booth <rick@itron-ca.com>
Date: Wed, 20 Nov 96 11:48:14
To: hfsig@tapr.org
Message-ID: <Chameleon.848519590.rick@rickb.itron-ca.com>
Rendered body type: text/plain

In spite of the fact the sampling process at the output of an IF
can be performed at rates consistent with the BANDWIDTH there are
certain s/h (and/or a/d) considerations that must account for the
IF frequency being used, the most notable being the aperture
jitter. Aperture jitter translates into phase noise and is worse
as the IF frequency is increased. This is easily observed in a
constellation plot at the output of a digital radio...good s/h's
with low jitter look clean and lousy s/h's look noisy...no free
lunch.
Rick

W6NZK

Rick Booth
'95 900SS SP

E-mail: rick@itron-ca.com
----

46. Re: [HFSIG:1730] RE: S/H

From: k6sti@n2.net (Brian Beezley)
Date: Wed, 20 Nov 1996 15:34:31 -0800
To: hfsig@tapr.org
Message-ID: <199611202334.PAA22647@ravel.n2.net>
Rendered body type: text/plain

>In spite of the fact the sampling process at the output of an IF
>can be performed at rates consistent with the BANDWIDTH there are
>certain s/h (and/or a/d) considerations that must account for the
>IF frequency being used, the most notable being the aperture
>jitter. Aperture jitter translates into phase noise and is worse
>as the IF frequency is increased. This is easily observed in a
>constellation plot at the output of a digital radio...good s/h's
>with low jitter look clean and lousy s/h's look noisy...no free
>lunch.
>Rick
>W6NZK
>------------------------------------
>Rick Booth
>'95 900SS SP
>E-mail: rick@itron-ca.com
>------------------------------------
>
>
>

I was wondering why the new radios with IF DSP use an 11-kHz or something
last IF. Seemed crazy, but must be the S/H problem.

Brian Beezley, K6STI
k6sti@n2.net

47. Re: TU-170

From: k5cnf1@juno.com (Richard B Morrow)
Date: Wed, 20 Nov 1996 06:58:35 PST
To: hfsig@tapr.org
Message-ID: <19961120.070032.8279.0.k5cnf1@juno.com>
Rendered body type: text/plain

>Take a look at the filtering after the mark/space detector and see if
>the
>output transitions are restricted to a slower rate. You may have to
>decrease the value of one or more filter caps to get the higher data
>rates.
 Can you explain what I am going to look for? I got to thinking about it
and realized that I just might not have a clue... which is not unusual
at times.

>There are several packages that run RTTY from a PC that would require
>no mod to the TU-170 at all - except for perhaps an adapter to get the
>RS-232 levels conditions for your PC. I don't recall if the I/O of
>the TU-170 is RS-232 compatable or just TTL - but you probably have all
>the specs there in your docs.

this ne does not have the RS-232 output but I think that there were some mods to add that in the
documentation I have.. Just have to look for it I
guess

>I have used an Amtor freebie program on HF with some success, and the
>TU-170 should work well on Amtor with the Baud filter modified to pass
>the 100 Baud data. On the other hand - if the Flesher specs say it was
>designed to pass 100 Baud RTTY signals - then no modification would be
>required for Amtor.

That would be a real deal if I could find it. My Rig is a TS-430 and I
can whip up just about anything to interface it. I am busy with repairs
on my motorcycle and car, Plus I am doing voulinteer work at the local
museum. Don't know if you have been seeing anything in your local paper
about the shipwreck in Matagorda Bay, The La Belle, which was one of the

La Salle expedition ships that went aground in 1686 or so. The cannon
that came up from that wreck was a real beauty. BUT I digress. I am doing the electronics for the
reverse electrolysis that gets the salt out of
the metal out of whatever they pull out out of the slat water.

I figure that the TU 170 could be used for just about anything with the
right program was to be had.

I do appreciate your input to the smoke testing that will be conducted
later when I get all of the stuff hooked up.. By the way what bands are you on any way I can be found on
 7.213 at times, but right now I am sort of off the air while I move furnature in the shack so I can
have anothe
desk to pile more stuff on and cover everything up even better..

73, Richard, K5CNF

>There is a program that can be purchased to run Pactor to an external
>modem also - which would require modifying the TU-170 Baud fliter to
>pass
>the 200 Baud Pactor signals. I do not recall the name of the program
>-
>but I have worked several hams using it.
>
>Look on the QRZ CD Rom for programs that run on the PC to run various
>modes. You may have some ham buddies who have that CD who can help.
>
>If you have trouble locating any of the specific programs - let me
>know
>and I will see what I can dig up.
>
>73 - Jim - w5zit@juno.com
>
>

48. Re: [HFSIG:1732] Re: TU-170

From: Lyle Johnson <lylej@azstarnet.com>
Date: Wed, 20 Nov 1996 21:56:24 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19961120215749.00371cb8@pop.azstarnet.com>
Rendered body type: text/plain

Richard,

>That would be a real deal if I could find it. My Rig is a TS-430 and I
>can whip up just about anything to interface it.

N7CL wrote up some mods that you'll want to do to your TS430S if you plan on
using it for AMTOR or any other fast switching mode between receive and
transmit. It was in PSR several years ago. I did it to my '430 back when I
owned one and it made the difference between being a useful rig for AMTOR
and a not very useful rig.

Cheers,

Lyle

49. Re: [HFSIG:1733] Re: TU-170

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Wed, 20 Nov 1996 23:45:36 -0600
To: hfsig@tapr.org
Message-ID: <v03007806aeb99cbd4ec7@[128.83.176.153]>
Rendered body type: text/plain

The index of articles is on the web page under PSR link on the top page
http://www.tapr.org It is in XLS, CSV, and TXT formats. Should be easy
enough to search and locate.

Cheers - Greg, WD5IVD

>Richard,
>
>>That would be a real deal if I could find it. My Rig is a TS-430 and I
>>can whip up just about anything to interface it.
>
>N7CL wrote up some mods that you'll want to do to your TS430S if you plan on
>using it for AMTOR or any other fast switching mode between receive and
>transmit. It was in PSR several years ago. I did it to my '430 back when I
>owned one and it made the difference between being a useful rig for AMTOR

>and a not very useful rig.
>
>Cheers,
>
>Lyle

Greg Jones, WD5IVD
Austin, Texas
wd5ivd@tapr.org
http://www.tapr.org/~wd5ivd
-----

50. Re: [HFSIG:1734] Re: TU-170

From: Lyle Johnson <lylej@azstarnet.com>
Date: Thu, 21 Nov 1996 08:00:17 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19961121080143.0030bb7c@pop.azstarnet.com>
Rendered body type: text/plain

At 01:56 AM 11/21/96 -0600, you wrote:
>The index of articles is on the web page under PSR link on the top page
>http://www.tapr.org It is in XLS, CSV, and TXT formats. Should be easy
>enough to search and locate.

Yep! Jan '86, Issue 18, page 11.

51. Spanish Article about EVM56002

From: "Eduardo Jacob" <jtpjatae@bicc00.bi.ehu.es>
Date: Thu, 21 Nov 1996 19:41:36 +0000
To: hfsig@tapr.org
Message-ID: <9611211944.AA38665@bicc00.bi.ehu.es>
Rendered body type: text/plain

I have just setup a page with an article written by Jabi EA2ARU
and me. It is inspired in Johan's one and is going to be published
in a Spanish Amateur Magazine (I think) in December. It mentions
the work of several people and it is meant to be a contribution
to the spanish speaking ham comunity in order to help them beginning
to play with the EVM. I think there will be another soon.

        The article is meant to be a living article so I welcome any
comments about it. It is located at

        http://bipt.bi.ehu.es/~jtpjatae/articulo.html

        These pages are my first ones, so contributions and comments are
welcome.

        Eduardo EA2BAJ

Eduardo Jacob - Area de Ingenieria Telematica
Departamento de Electronica y Telecomunicaciones
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UPV / EHU   Fax: +34-(9)4-441 4041
Alda Urquijo s/n   E-mail: jtpjatae@bi.ehu.es
E-48013 - Bilbao (Spain)   : 100021,2212 Compuserve
Ham: EA2BAJ   VHF PACKET: EA2BAJ @ EA2URV

52. Re: [HFSIG:1720] Sigma Delta

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Thu, 21 Nov 1996 21:26:52 -0500
To: hfsig@tapr.org
Message-ID: <32950F6C.19D4@worldnet.att.net>
Rendered body type: text/plain

Alan Bloom wrote:
>
> I promised to try to find a reference explaining Sigma-Delta A/D converters.
> None of my DSP textbooks seem to cover the topic. I did find an app note
> from Harris Semiconductor (AN9504, May 1995) that does a pretty good job.
> I can copy it if anyone is interested.
>
> Basically a sigma-delta converter is formed thus:
>

What follows had all the elements, but hooked up incorrectly.
The one bit DAC output is integrated for a very short period of
time. The output of the integrator is fed into the summer
where it is subtracted from the Analog Input. This goes into
the comparator where the Digital output comes and the error
signal tells the DAC whether over the next period of integration
it should output plus or minus voltage. Now it should make more
logical sense to you as it is doing proportional guidance with
bang bang control if you want to go look at an old control
theory book. This is the simplest sigma-delta ADC. They get
much fancier.

Bob

****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio. Scouts: Council Commissioner
*
* East Windsor, NJ 08520   * Post 995 Advisor, Troop 5700 Advancement,
*
* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061   * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240   * Council. Brownie Troop 193
*
****

53. Re: Sigma Delta

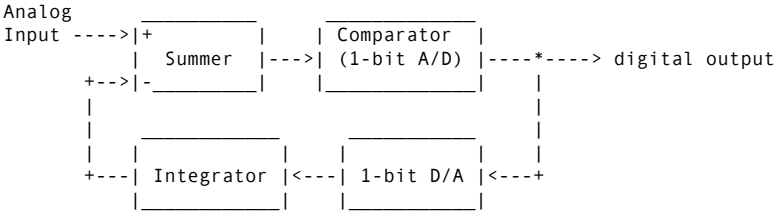
From: Alan Bloom <alanb@polecat.sr.hp.com>

Date: Fri, 22 Nov 1996 12:30:04 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611222030.AA082054604@polecat.sr.hp.com>
Rendered body type: text/plain

Robert McGwier <rwmcgwier@worldnet.att.net> wrote:

>Alan Bloom wrote:
>>
>> I promised to try to find a reference explaining Sigma-Delta A/D converters.
>> None of my DSP textbooks seem to cover the topic. I did find an app note
>> from Harris Semiconductor (AN9504, May 1995) that does a pretty good job.
>> I can copy it if anyone is interested.
>>
>> Basically a sigma-delta converter is formed thus:
>>
>
>What follows had all the elements, but hooked up incorrectly.
>The one bit DAC output is integrated for a very short period of
>time. The output of the integrator is fed into the summer
>where it is subtracted from the Analog Input. This goes into
>the comparator where the Digital output comes and the error
>signal tells the DAC whether over the next period of integration
>it should output plus or minus voltage. Now it should make more
>logical sense to you as it is doing proportional guidance with
>bang bang control if you want to go look at an old control
>theory book. This is the simplest sigma-delta ADC. They get
>much fancier.

I think what you're describing is this:



That is actually a "Delta" converter. Compared to a Sigma-Delta
converter it has the disadvantage that there is a slew rate limit
on the input signal. A high-amplitude, high-frequency analog input
will exceed the slew rate capability of the integrator, causing
severe distortion. That's not too serious a constraint for codec
applications, because most of the energy in the human voice is at
low frequencies. The high-frequency energy is low enough not to

exceed the maximum slew rate.

It's called a "Delta" converter because the digital output represents
the rate of change (delta) at the input. A continuous DC input voltage
causes the digital output to represent zero volts (+1 half the time,
-1 half the time). To recover the audio, you have to integrate the
output (run it through a -6 dB per octave filter).

Another way to view the slew-rate problem is to note that the digital
output represents the slope (time rate of change) of the input signal.
The slew rate limit is the point where the output is all +1's or -1's.

In a Sigma-Delta converter, the integrator is moved from the input
to the output of the summer. That solves the slew rate problem and
gives a flat output frequency response.

Alan Bloom N1AL

54. Re: Sigma Delta

From: Rick Whiting <rwhiting@winternet.com>
Date: Fri, 22 Nov 1996 18:12:32 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <199611230012.SAA09966@icicle.winternet.com>
Rendered body type: text/plain

There is a discussion of sigma-delta converters in:

   Pohlmann, K.C. (1991). "Advanced Digital Audio." Carmel, IN: SAMS

e.g., p. 407-408, 413-414. For example, there is an internal block diagram
of a DSP56ADC16 sigma-delta A/D converter on p. 413.

Regards/ Rick W0TN

  Richard A. (Rick) Whiting   Home Phone: + 1 612 550 1213
  RF Engineer, AirTouch Cellular   Work Phone: + 1 612 595 5065
  5780 Rosewood Ln. N.   E-mail: rwhiting@winternet.com
  Plymouth, MN 55442-1411   Packet: W0TN @ WB0GDB.MN.USA.NOAM
 ----

55. Re: [HFSIG:1726] RTTY

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 25 Nov 1996 21:42:02 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611260542.VAA27906@servo.qualcomm.com>
Rendered body type: text/plain

>In designing the receiver, you wouldn't go far wrong to assume that
>there is no transmit filter and that each mark or space tone is held
>for the entire duration of the symbol as Phil illustrated above. The

If this is true, then the transmitted spectrum would have a "comb"
(sin(x)/x) shape and the simplest matched filter would be an
integrate-and-dump.

That is, if you can form a local estimate of the mark and space tones,
simply multiply the incoming signal separately by the local mark &
space estimates, square each to compute power, subtract one from the
other and integrate the result over the bit interval. Sample at the
end of the bit interval and you have your decision variable.

In other words, integrate

        cos(2*pi*fmark*t)*s(t))**2 - cos(2*pi*fspace*t)*s(t))**2

over the bit interval. If the result is positive, you have a mark. If
negative, you have a space. Or you can use the numerical result as a
soft decision sample for a Viterbi decoder. Dump the accumulated value
and start again at 0 for the next bit.

This is a classic optimum noncoherent demodulator, with the signal
pulse shaping omitted because it's square to begin with. Note that
you don't need knowledge of mark & space phase, only their frequencies,
and then only accurately enough so that the phase "drift" over a bit
interval is acceptably small. The faster the data rate, the sloppier
your frequency estimate can be.

Phil

56. Re: [HFSIG:1740] Re: RTTY

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Tue, 26 Nov 1996 03:39:39 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611261139.DAA02000@unix.ka9q.ampr.org>
Rendered body type: text/plain

A correction. The optimum noncoherent demodulator should integrate
the following over the bit interval:

```
        cos(2*pi*fmark*t)*s(t))**2 + sin(2*pi*fmark*t)*s(t))**2
        - [cos(2*pi*fspace*t)*s(t))**2 + sin(2*pi*fspace*t)*s(t))**2 ]
```

That is, the mark detector should produce the sum of the energies
received in both the in-phase and quadrature channels with respect
to the mark carrier reference, and similarly with the space detector.

The decision variable is, as before, the difference between the total
energies seen in these two channels over the bit interval.

By the way, the matched filter detector can be implemented in one
of two ways.

The first way is to sample the output of a bandpass filter whose
response is the complex conjugate of the signal spectrum. "Complex
conjugate" simply means that the amplitude response is the same, and
the phase response is adjusted to correct for any frequency dispersion
in the signal.

The second way is to multiply the incoming signal by a local
time-reversed replica of the signal's nominal pulse shape, integrate
over a bit interval, then sample. (If the nominal pulse shape is
symmetrical, then the time-reversal isn't noticeable).

These two ways are in fact exactly equivalent. This is most easily
seen when you use a FIR (finite impulse response) filter to implement
method #1, as it does exactly the same thing as method #2 --
multiplying the incoming signal by the time-reversed replica of the
nominal pulse shape and summing (integrating) the results.

Phil

57. [HFSIG:1741] Re: RTTY

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 26 Nov 96 13:37:55 +0000
To: hfsig@tapr.org
Message-ID: <H000029202dadbaf@MHS>
Rendered body type: text/plain

> A correction. The optimum noncoherent demodulator should integrate
> the following over the bit interval:
>
> cos(2*pi*fmark*t)*s(t))**2 + sin(2*pi*fmark*t)*s(t))**2
> - [cos(2*pi*fspace*t)*s(t))**2 + sin(2*pi*fspace*t)*s(t))**2]
>
> That is, the mark detector should produce the sum of the energies
> received in both the in-phase and quadrature channels with respect
> to the mark carrier reference, and similarly with the space detector.

This sounds better, I was a bit confused by your pervious post. I have
implimented something similar in BTL for the 'Narrow' mode. But instead
of summing the squares of I and Q, I simple take the magnitude of I+jQ,
and compare the magnitudes. This would not be as good for a soft
decision variable, but it has no effect on the result of a hard decision
comparison between mark and space, ie is |A|>|B| then A*A > B*B. Since
RTTY has no way of detecting errors, this seems optimal to me.

However in actual practice, I find this method is rarely any better than
my non-optimnal detection, which simply uses +/- 100Hz filters (+/-170Hz
nulls) and takes the amplitude, then does the bit period intergration.
My overall receiver filter impulse is fairly rounded, complete with ISI,
and suffers from amplitude detection between the filter and the bit
period integrator, but still, it actually appears to work just as well
as the optimum solution, better if practicalities of tuning, listening
to several stations with slightly different frequencies, and the
170/200Hz shift uncertianty, are considered.

I suspect the lower tollerance to tuning of the optimal detection method

means that if the optimal method is slightly off tune, by say 20Hz, then
you have lost the few dB you might have gained, where the non-optimal
solution can work OK +/-75Hz off, looseing a few more dB of course, but
less than you would have lost with narrower filters.

If the optimal solution is more sensitive, in practice it makes little
difference given the fading nature of the channel, and the tuning
uncertainties. I have only ever got a significant improvement in
performance using the optimal filters when there as another RTTY signal
on the same frequency (shifted by maybe 50Hz) where I was able to filter
it out. Of course its very rear that this can be done.

Cheers,

Rob

58. Re: [HFSIG:1742] Re: RTTY

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 26 Nov 1996 23:41:54 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199611270741.XAA01372@servo.qualcomm.com>
Rendered body type: text/plain

>This sounds better, I was a bit confused by your pervious post. I have
>implimented something similar in BTL for the 'Narrow' mode. But instead
>of summing the squares of I and Q, I simple take the magnitude of I+jQ,
>and compare the magnitudes. This would not be as good for a soft

"Taking the magnitude of I+jQ" normally implies taking the sum of the
squares of the real and imaginary components, but from the later
context I assume you mean simply summing the amplitudes of the real
and imaginary components without any squaring operation?

If so, this is suboptimal even in an uncoded hard-decision
situation. Consider the following example in which your demodulator
algorithm makes an error while the optimal one does not:

Assume the mark channel integrator gives the complex result .707 +
.707j while the space channel integrator gives the result 1.3 +
0.0j. The space channel clearly has the stronger signal with magnitude
sqrt(1.3**2 + 0**2) = 1.3 as compared to the mark channel's
sqrt(.707**2 + .707**2) = 1.0. Yet your algorithm would give a
magnitude of 1.414 for the mark channel, causing it to be chosen in
error over the space channel.

Phil

59. Re: RTTY

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 28 Nov 96 10:00:04 +0000
To: hfsig@tapr.org
Message-ID: <H000029202dff828@MHS>
Rendered body type: text/plain

> Assume the mark channel integrator gives the complex result .707 +
> .707j while the space channel integrator gives the result 1.3 +
> 0.0j. The space channel clearly has the stronger signal with magnitude
> sqrt(1.3**2 + 0**2) = 1.3 as compared to the mark channel's
> sqrt(.707**2 + .707**2) = 1.0. Yet your algorithm would give a
> magnitude of 1.414 for the mark channel, causing it to be chosen in
> error over the space channel.

Sorry, yes, I agree. Perhaps I didn't explain clearly enough. The
amplitude I use is the magnitude of the complex vector ie sqrt(I*I +
Q*Q), although in actual practice I use a much faster approximation
which is accurate to about 1%. If I were to use squares I would need to
use 32 bit words or floating point, and there is no advantage for hard
decisions. I can always square the magnitude later if needed, and ovoid
carrying around 32 bit or floating point variables.

Cheers,
Rob

60. Re: RTTY

From: Marijan Miletic <s56a@s55tcp.ampr.org>
Date: Thu, 28 Nov 96 12:05:27 EST
To: hfsig@tapr.org
Message-ID: <20383@s55tcp.ampr.org>

Rendered body type: text/plain

Following my previous KISS postings, it is well known that IQ power can be also
calculated as sum of larger signal with 0.4 of smaller. I use right shift for
1/2 of smaller signal...
73 de Mario, S56A, N1YU.

61. [HFSIG:1745] Re: RTTY

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 28 Nov 96 18:20:23 +0000
To: hfsig@tapr.org
Message-ID: <H000029202dff83a@MHS>
Rendered body type: text/plain

> Following my previous KISS postings, it is well known that IQ power
> can be also calculated as sum of larger signal with 0.4 of smaller. I
> use right shift for 1/2 of smaller signal...

Sounds nice an simple!

Here's my one

I=abs(I)
Q=abs(Q)
if (Q>I) swap I and Q

If (Q>I/2) M = I*7/8 + Q*9/16
else M= I + Q/4

Shift and adds are used for the fractions.

My ASM code for this takes 0.9uS on my 486DX-2 / 80MHz

For phase:

phase = table[Q/(I+Q], for 0-45 degrees.

256 entries in table for 45 degrees

The Q/(I+Q) is an approximate phase to get reasonable linearity for the
table index, and the table values are 16 bits, used as a correction
table. Thus there are 256*8=2048 discernible phases.

I haven't done timing for this, but I expect it is faster than the ABS
function, since there are no shift and adds, but one multiply.

Cheers,

Rob

December 1996

Original source file: hfsig.9612
Source: https://archive.org/details/tapr-1997
Recovered from: https://archive.org/download/TAPR-HF_1996-12/TAPR-HF_1996-12.txt

1. Advantages of SS

From: Gwyn Reedy <gwyn@paccomm.com>
Date: Thu, 1 Aug 1996 06:38:51 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB7F74.1B3654E0@gwyn.paccomm.com>
Rendered body type: text/plain

Phil Karn said (in conclusion to an excellent technical tutorial)

>(SS has) he desirable effect of uniformly distributing the load over the band. The law of large
>numbers kicks in and the carrying capacity of the band increases thanks to the improved ability to >dynamically share spectrum.

>And this is ultimately why we should encourage spread spectrum on all of our bands, especially >the most crowded ones.

I think this encapsulates the essence of SS resistance. Many amateur spectrum users (might I call out contesters and DXers) do not want to share spectrum; their satisfaction with the hobby derives from maximizing their personal communication effectiveness both on transmit and receive. Sharing is not one of their concerns. And this is the kind of person the ARRL considers the mainstream of amateur radio.

Gwyn Reedy, W1BEL
PacComm Packet Radio Systems
http://www.paccomm.com

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2. Re: [HFSIG:1398] Advantages of SS

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 1 Aug 1996 05:51:32 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608011251.FAA18796@servo.qualcomm.com>
Rendered body type: text/plain

Gwyn's comments are perhaps a bit blunt, but he's certainly on the right track.

Another issue I've been meaning to comment on is the relative efficiency of relay vs direct communications. The near-far problem comes up mainly in the context of a station desiring to communicate to a very distant point, typically requiring high power, while nearby SS stations are communicating among themselves at relatively low power.

I believe Robert Glassey made an earlier comment along these lines when he said that SS communications assume local cellular-style communications. And he was quite right, if by "cellular style" he meant heavy geographical reuse of spectrum, with relaying rather than high power used to reach more distant stations. "Cellular style" does NOT mean you're unable to talk DX, you just may not be able to do it directly. (I routinely "work" hundreds of miles with my little 300mw handheld 890 MHz unit...)

This strikes at another major aspect of ham radio: the "self-sufficient ethic". Despite the literal meaning of the third letter in "ARRL" the amateur culture strongly favors direct end-to-end communications, especially in the HF bands. Repeaters are generally not allowed on these bands, and even where they're allowed (e.g., on 10m) they are usually excluded from most contests as they're considered to be "cheating".

Now some of this undoubtedly comes from the fact that historically, repeaters consume more spectrum than simplex and are also limited resources that are better reserved for other things, like mobile users and emergencies. But there is also an underlying theme of self-sufficiency that says a ham should have to do it all by himself to gain credit. Big antennas and massive linear amplifiers command a lot of respect, while resorting to a repeater is at least an admission of inferiority, or even downright dishonest (if used in a competition).

Spread spectrum turns this culture on its head. Not only does the
average ham have to accept the obviously absurd notion that a wideband
signal can be more spectrally efficient than a narrowband signal, but
he is also being asked to believe that repeaters can be more
spectrally efficient than simplex! That is, a network of relatively
low power store-and-forward spread spectrum packet radios with
min-energy routing is far more spectrally efficient than direct high
power end-to-end operation.

Even the reliability aspect of not wanting to depend on a repeater
(sometimes cited in the context of emergency communications) is really
not an issue here. All you need are enough stations to back each other
up and you can have a network that, overall, is much *more* reliable
than DX simplex thanks to the shorter inter-node propagation paths.

All this has actually always been true even for narrowband modulation;
the near-far aspect of spread spectrum just makes it more glaringly
obvious. With cooperative relaying, it would still be possible to
work DX on HF. You just wouldn't necessarily do it
directly. Unfortunately, for a lot of hams this wouldn't be the
same.

Phil

3. Shared SS

From: k6sti@n2.net (Brian Beezley)
Date: Thu, 1 Aug 1996 07:32:11 -0700
To: hfsig@tapr.org
Message-ID: <199608011432.HAA07242@ravel.n2.net>
Rendered body type: text/plain

>But when the band gets loaded, and especially when the typical traffic
>consists of a lot of short, bursty transmissions, spread spectrum in
>general and frequency hopping in particular has the desirable effect
>of uniformly distributing the load over the band. The law of large
>numbers kicks in and the carrying capacity of the band increases
>thanks to the improved ability to dynamically share spectrum.

Narrowband users already dynamically share spectrum. They QSY to avoid QRM.
But if their frequencies were overlayed with SS, they would experience QRM
they couldn't evade.

Any attempt to mix SS and narrowband modes on HF will have to address the
following complaint from day one:

Before this thing you call spread spectrum came along, I could always have a
nice chat on 20-meter SSB. If QRM arose, we either convinced the other guy
to move or we moved ourselves. But now, whenever I listen on the band I
hear funny noises in the background. I can still hold a conversation but
it's not pleasant listening to this stuff. I tried QSYing to find a clear
spot but it didn't help--the damn stuff is everywhere. I missed a new
country last week when that computer brat down the street fired up on SS and
took out the whole band for hours. When the SS guys across town get on, I
can hear well enough to check into the Old Farts net, but I often miss my
turn when net control fades into their bleeps and bloops. Frankly, I don't
get on 20 much any more because it's just not worth the trouble. I want my
band back.

Any technical argument for SS that can't respond simply and directly to a
complaint like this will be immediately dismissed as BS. Shared SS won't
happen until hundreds of thousands of hams like the one imagined above are
convinced that SS will bring benefit instead of destroying their present ham
activities.

Brian Beezley, K6STI
k6sti@n2.net

4. Re: [HFSIG:1400] Shared SS

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 1 Aug 1996 10:51:17 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608011551.KAA23596@www.kelly-afb.org>
Rendered body type: text/plain

In your message you write:
>
> Narrowband users already dynamically share spectrum. They QSY to avoid QRM.

> But if their frequencies were overlayed with SS, they would experience QRM
> they couldn't evade.
>
> Any attempt to mix SS and narrowband modes on HF will have to address the
> following complaint from day one:

        Way back there in my AFRes days, we were running SSB on several
        channels in the 8.4 - 8.9 MHz and in the 11.5 MHz range and the spooks
        about 300 meters were running SS between 3 -15 MHz to the same
        locations that we were. Propagation to the location we were both
        talking to was fair on 8 MHz and poor on 11 Mhz with our equipment.
        The spooks had very good communications. We were running 500 Watts
        PEP and they were running less that a 100 watts IAW their specs.
        Also, it didn't matter if they were on-the-air or not, it had no
        affect on our communications and I don't recall seeing an
        increase in channel noise...it darn sure didn't bother the marine
        CW station on the channel ajacant to our main channel (frequency).
        Also, there was another spook station located about 5 miles away
        and we couldn't tell when either was transmitting unless we called
        them on the fieldfone and asked. I might also mention that they
        were running broadband antennas and we were running dipoles so our
        antennas were better.

>
> Before this thing you call spread spectrum came along, I could always have a
> nice chat on 20-meter SSB. If QRM arose, we either convinced the other guy
> to move or we moved ourselves. But now, whenever I listen on the band I
> hear funny noises in the background. I can still hold a conversation but
> it's not pleasant listening to this stuff. I tried QSYing to find a clear
> spot but it didn't help--the damn stuff is everywhere. I missed a new
> country last week when that computer brat down the street fired up on SS and
> took out the whole band for hours. When the SS guys across town get on, I
> can hear well enough to check into the Old Farts net, but I often miss my
> turn when net control fades into their bleeps and bloops. Frankly, I don't
> get on 20 much any more because it's just not worth the trouble. I want my
> band back.
>
> Any technical argument for SS that can't respond simply and directly to a
> complaint like this will be immediately dismissed as BS. Shared SS won't
> happen until hundreds of thousands of hams like the one imagined above are
> convinced that SS will bring benefit instead of destroying their present ham
> activities.
>
>

        Based on my experience (see above) I have to disagree.

> Brian Beezley, K6STI
> k6sti@n2.net
>
>
        Walt/K5YFW

```
==========================================================================
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development.      |
==========================================================================
|                                  |                                     |
|                                  | The greatest dangers to liberty     |
| Walt DuBose - K5YFW              | lurk in insidious encroachment      |
| E-Mail k5yfw@www.kelly-afb.org   | by men of zeal, well-meaning        |
| Business Telephone: (210)925-6081 | but without understanding.         |
|     Home Telephone: (210)696-3196 |                                    |
|                                  |   - Justice Louis D. Brandeis       |
|                                  |                                     |
==========================================================================
```

5. Wide band modes

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 1 Aug 96 17:42:08 +0100
To: hfsig@tapr.org
Message-ID: <H000029202272ac5@MHS>
Rendered body type: text/plain

Hi Phil,

> Error correction is an especially powerful technique with frequency
> hopped spread spectrum, as opposed to direct sequence. Both the
[snip]

> But on a Rayleigh fading channel, both coherent and noncoherent
> schemes exhibit a error rate that decreases only *linearly* with
> increasing SNR. That is, if you want to decrease your bit error rate
> by a factor of 10, you need to increase your power by a factor of
> ten. The different schemes do differ in the precise amount of power
> required (e.g., ideal coherent BPSK is 3 dB better than DBPSK, which
> is 3 dB better than BFSK) but they are all exactly parallel over a
> very wide range of SNR.
>
> Some illustrative figures: for a BER of 1%, uncoded BPSK on a Rayleigh
> channel requires an Eb/N0 of about 20 dB. To lower the BER to .1%, you
> need 30 dB. To achieve 10^-6, you need an coax-melting Eb/N0 of 60dB!

> If you're after a low error rate on a fading channel, the proper use
> of coding and interleaving can have a dramatic effect. While coding
> can get you perhaps 7 dB at best on an AWGN channel, on a Rayleigh
> fading channel it's not hard to achieve as much as 35 - 40 dB in power
> savings compared to uncoded transmission. That's pretty impressive.

It is indeed. I have no arguments here. I've long suspected as much, although I've never had the numbers that show it. Although even very simple ECC can make back most of this 35-40db difference. ie and AMTOR ARQ works well with a BER of 10-2 (1%), as does RTTY where the operator fills in the errors. I think BPSK requires about 4 or 5dB for 1% BER in a AWGN channel, so were talking about 15dB extra power over the non-fading case. I guess most of this need for power is to give enough fade margin so that the signal only drops below the threshold 1% of the time.

I guess without any extra power, the signal would be above the threshold 50% of the time. Thus we have a 50% BER. A 1/2 rate ECC would ideally deal with this, although I guess a bit more redundancy would be usefull.

In the light of this 15dB fade margin, I wonder how much effect soft decoding would really have. Would erasure decoding be just as good? What I'm geting at is PACTOR II's K=9 or is it 11 contraint length. Would k=5 be just as good? Or could we just go for plain Reed Soloman coding? This would make PC implementation of an HFSIG protocol more realistic.

But getting back to your spreading argument, I've just suggested trippling the bandwidth to overcome fading, or have I?

I recall Ungerboch (I can never spell that) said that the solution was to go for multi levels, even if you cannot actually resolve each one discretly. Enough extra bits get through to give you the ECC bits for free (power and bandwidth-wise). This works in the AWGN channel, but in the fading channel it could be a real bonus, since some of the time the signal is several times what is required, allowing many additional levels. I imagine special attention would have to be paid to equalisation to overcome ISI when phase levels of 16+ PSK are used. (I've done on air measurements of PACTOR II transmissions using 8-psk, and the phase coherence is remarkably good over the DL-G path on 14MHz). Even amplitude levels could be considered since for a significant proportion of the time (>90%) they could be resolved. 64-QAM may not be unrealistic!! - giving an actual QPSK data rate, while overcomming the fading, and allowing high data rates when conditions are good. PACTOR II and clover seem to do the reverse, reducing ECC as level increase. Ungerboch says this is the wrong way around.

> Now these gains are admittedly somewhat separate from the issue of
> spread spectrum frequency hopping. But some bandwidth expansion just
> from the coding is inevitable, and this expansion is likely to be
> greater than on the AWGN channel.

Not inevitible, but I can still see that a wider band signal will have faster fades that are more easilly dealt with. I think the wider bandwidth helps when you consider interference from other users. Still a narrow a bandwidth allows more users in the same wide bandwidth without interference. 6 of one, half dozon of another as they say. I dont think either can make full use of the bandwidth.

> But on the Rayleigh fading channel, both the achievable coding gains
> and the bandwidth expansion ratios are considerably greater. Not only
> are orthogonal signalling constellations with a fairly large number of
> dimensions necessary (e.g., M-ary FSK with large M), but the optimum
> FEC code rates for use with these schemes are lower than on the AWGN
> channel. For example, 16-ary FSK requires 4 times as much bandwidth as
> BFSK for the same data rate, and rate 1/8 coding is optimal for this

> value of M. That's a 32x bandwidth expansion, but it performs about 5
> dB better than a rate 1/2 code combined with BFSK (which only doubles
> the uncoded bandwidth).

Hmm, this is interesting, expecially the 5dB increase over the BPSK
case. I wonder what it would be over the 64-QAM case.

So we are now talking about say 500Hz x 32 or about 16kHz bandwidth.
Interesting. Of course this is not hopping yet.

> And it may not always be necessary to spread at all; on a lightly
> loaded band, you could simply turn off hopping once you've found what
> appears to be an empty channel. (This is equivalent to dropping every
> channel but one from the hopping list).

How resistant would this M-ARY scheme be to other signals? I wonder if
different convolution codes could be used to provide further resistance
between signals (getting very spread spectrum like now), with some
sequential soft decoding being used to distinguish between the different
stations, rather than just using a majority vote detector. Could this
provide enough resistance to other stations on the same channel? It may
be possible to decode 100% a weaker station while a much stronger
station, or two use the same channel. MARY FSK seems to have quite an
advantage here.

Hopping inaddition to this would allow for more stations again.

I'm actually fairly well convinced that with a properly designed digital
radio, with a wide bandwidth and demod controlled AGC could overcome the
near far problem quite well, especially if spread enough so that local
impulses cause a BER of < 10%. I even like the idea!

The concern is about sharing the band with other users. I dont think
there is much concern about sharing the band with other wideband users,
and I can see that it may well be a more efficent way of using the
allocated spectrum, but sharing in an amateur enviroment is not just
about getting the maximum number of user into a particular band. Amateur
radio must allow many different modes, each with different strengths and
weakness to co-exist without causing each other interfence. The amateur
service is an 'experimental' service, which allows people to 'play' with
radio techniques and it many different forms to learn more about radio,
and have fun doing it.

If this mode can be made to work in a unspread (25kHz) bandwidth (same
25kHz allocation world wide) then I would support it, as a start, and if
it proves so popular that spreading becomes nessesary to accomodate the
number of stations, then there may be a good argument for allocating say
100KHz to the mode allowing hopping, without seriously effecting other
hams and their legitimate interests. If demand further increases then we
may have a good case for a NEW BAND, especially as HF spectrum is being
slowly cleared of commercial users.

I would support all these cases, but I could never support turning over
whole existing bands to spread spectrum. Its not in the spirit of ham
radio.

I posted a list of various assumptions that are made when arguing the
case for spread spectrum. Most of them still stand, although I think we
have improved the near-far problems.

1 & 4. Hams do want to work DX, simplex. Its not selfish, its the spirit
of ham radio, its what excites many people about the hobby. Maybe when
the bands are so badly conjested that DX is impossible SS may be our
only hope, but that would reduce ham radio to a single mode, with little
advantage (if any) over commercial celular data networks and the
internet. I think that day would be an especially black one for the
furure of the hobby, and many would feel ham radio is not worthwhile
anymore.

2 & 3. The near far thing. Still a problem for non-fec data users, and
SSB operators without DSP based noise blankers. They still have the
right to experiment with less than perfect modes, everyone must start
somewhere, and simple modes are a good starting point.

5. Ham radio is an experimental service for self education. Reducing ham
radio to a single mode, with only the prospect of contact with your
local ISP, or someone only a few 100 miles away being considered DX,
would pretty much spell the end of amateur radio. If we can only use it
as the poor mans data network, then the rich men will get it.

Maybe this is the future of ham radio, but I'd like to think otherwise.

Rob.

6. Re: [HFSIG:1400] Shared SS

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Thu, 1 Aug 1996 13:38:36 -0400
To: hfsig@tapr.org
Message-ID: <200EC2C0.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

I'd like to know a little more about this 'computer brat down the
street.' How do you know he is using SS? What is his output power if
he is using SS? When the SS guys across town get on the band, how much
power are they using? Are you sure they are "SS guys?"

What do these 'funny noises' sound like? I'm not trying to be
condescending, Brian, but I think you are mistaking packet radio for
SS.

73s de
Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1400] Shared SS
Author: hfsig@tapr.org at BALT.SMTP
Date: 8/1/96 9:37 AM

>But when the band gets loaded, and especially when the typical traffic
>consists of a lot of short, bursty transmissions, spread spectrum in
>general and frequency hopping in particular has the desirable effect
>of uniformly distributing the load over the band. The law of large
>numbers kicks in and the carrying capacity of the band increases
>thanks to the improved ability to dynamically share spectrum.

Narrowband users already dynamically share spectrum. They QSY to avoid QRM.
But if their frequencies were overlayed with SS, they would experience QRM
they couldn't evade.

Any attempt to mix SS and narrowband modes on HF will have to address the
following complaint from day one:

Before this thing you call spread spectrum came along, I could always have a
nice chat on 20-meter SSB. If QRM arose, we either convinced the other guy
to move or we moved ourselves. But now, whenever I listen on the band I
hear funny noises in the background. I can still hold a conversation but
it's not pleasant listening to this stuff. I tried QSYing to find a clear
spot but it didn't help--the damn stuff is everywhere. I missed a new
country last week when that computer brat down the street fired up on SS and
took out the whole band for hours. When the SS guys across town get on, I
can hear well enough to check into the Old Farts net, but I often miss my
turn when net control fades into their bleeps and bloops. Frankly, I don't
get on 20 much any more because it's just not worth the trouble. I want my
band back.

Any technical argument for SS that can't respond simply and directly to a
complaint like this will be immediately dismissed as BS. Shared SS won't
happen until hundreds of thousands of hams like the one imagined above are
convinced that SS will bring benefit instead of destroying their present ham
activities.

Brian Beezley, K6STI
k6sti@n2.net

7. Re: [HFSIG:1399] Re: Advantages of SS

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Thu, 1 Aug 1996 13:44:54 -0400
To: hfsig@tapr.org
Message-ID: <200ED5B0.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

Phil,

Could you PLEASE run for president of the ARRL?

73s de
Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1399] Re: Advantages of SS
Author: hfsig@tapr.org at BALT.SMTP
Date: 8/1/96 8:05 AM

Gwyn's comments are perhaps a bit blunt, but he's certainly on the
right track.

Another issue I've been meaning to comment on is the relative
efficiency of relay vs direct communications. The near-far problem
comes up mainly in the context of a station desiring to communicate to
a very distant point, typically requiring high power, while nearby SS
stations are communicating among themselves at relatively low power.

I believe Robert Glassey made an earlier comment along these lines
when he said that SS communications assume local cellular-style
communications. And he was quite right, if by "cellular style" he
meant heavy geographical reuse of spectrum, with relaying rather than
high power used to reach more distant stations. "Cellular style" does
NOT mean you're unable to talk DX, you just may not be able to do it
directly. (I routinely "work" hundreds of miles with my little 300mw
handheld 890 MHz unit...)

This strikes at another major aspect of ham radio: the
"self-sufficient ethic". Despite the literal meaning of the third
letter in "ARRL" the amateur culture strongly favors direct end-to-end
communications, especially in the HF bands. Repeaters are generally
not allowed on these bands, and even where they're allowed (e.g., on
10m) they are usually excluded from most contests as they're
considered to be "cheating".

Now some of this undoubtedly comes from the fact that historically,
repeaters consume more spectrum than simplex and are also limited
resources that are better reserved for other things, like mobile users
and emergencies. But there is also an underlying theme of
self-sufficiency that says a ham should have to do it all by himself to
gain credit. Big antennas and massive linear amplifiers command a lot
of respect, while resorting to a repeater is at least an admission of
inferiority, or even downright dishonest (if used in a competition).

Spread spectrum turns this culture on its head. Not only does the
average ham have to accept the obviously absurd notion that a wideband
signal can be more spectrally efficient than a narrowband signal, but
he is also being asked to believe that repeaters can be more
spectrally efficient than simplex! That is, a network of relatively
low power store-and-forward spread spectrum packet radios with
min-energy routing is far more spectrally efficient than direct high
power end-to-end operation.

Even the reliability aspect of not wanting to depend on a repeater
(sometimes cited in the context of emergency communications) is really
not an issue here. All you need are enough stations to back each other
up and you can have a network that, overall, is much *more* reliable
than DX simplex thanks to the shorter inter-node propagation paths.

All this has actually always been true even for narrowband modulation;
the near-far aspect of spread spectrum just makes it more glaringly
obvious. With cooperative relaying, it would still be possible to
work DX on HF. You just wouldn't necessarily do it
directly. Unfortunately, for a lot of hams this wouldn't be the
same.

Phil

8. Re: [HFSIG:1404] Re: Advantages of SS

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 1 Aug 1996 13:59:16 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608011859.NAA24314@www.kelly-afb.org>
Rendered body type: text/plain

Hear...hear...

>
>   Phil,
>
>   Could you PLEASE run for president of the ARRL?
>
>   73s de
>   Tony, KE4ATO
>
>
> _____________________________ Reply Separator _____________________________
> Subject: [HFSIG:1399] Re: Advantages of SS
> Author: hfsig@tapr.org at BALT.SMTP
> Date: 8/1/96 8:05 AM
>
>
> Gwyn's comments are perhaps a bit blunt, but he's certainly on the
> right track.

9. Re: [HFSIG:1399] Re: Advantages of SS

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 01 Aug 1996 11:50:54 -0800
To: hfsig@tapr.org
Message-ID: <199608011902.MAA06926@PEAK.ORG>
Rendered body type: text/plain

Hi Folks,

I must say that I really enjoy the stimulating discussion about HF SS - extremely educational.

Personally, I'm not hearing convincing arguments either way, especially based on extreme situations or extrapolations.

Could someone please comment on the following:

1) If assumed that the center frequency of different DS SS operations are randomly chosen, what would that do to the near-far problem, i.e., if you use DS SS and experiencing interference from your neighbor DS SS operator, why not QSY out of harm's way of the offending signal envelope? This assumes that contacts are initially established on some known frequency. Am I right in assuming that if only two parties are involved in DS SS, that the PSD does have a maxima somewhere, even though it is a wideband, i.e, not quite white over the full bandwidth?

2) How many SS particpants does it take to start raising the noise floor to a point where it becomes objectionable? 2, 10, 100, 1000 ? If the number is 1000, would it be realistic, given a comparable narrow-band situation, to expect anyone to be able to conduct a QSO with that occupancy? Like a contest with 1000 RTTY operators calling simultaneously in the 14.090 - 14.065 20 meter segment.

As a lay person, and I admit I'm ignorant, the latter is the impression that I get from folks attempting to defend the narrow-band ideology. I'nt it time for someone to demonstrate a two-way digital QSO using SS on HF so we can all see for ourselves the magnitude of the real issues?

My two cents worth.

--Johan

10. Re: [HFSIG:1403] Re: Shared SS

From: Charles Brain <chbrain@dircon.co.uk>
Date: Thu, 01 Aug 1996 20:41:43 +0100
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19960801194143.0067b328@popmail.dircon.co.uk>
Rendered body type: text/plain

At 12:50 01/08/96 -0500, you wrote:
>   I'd like to know a little more about this 'computer brat down the
>   street.' How do you know he is using SS?

He isn't actually. In fact he is a member of Joe Public who has just turned his P.C on (The grey box with all the wires hanging out the back).

  :- }

- Charles

11. SS on HF bands

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Thu, 1 Aug 1996 13:31:39 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <199608012031.AA171931499@polecat.sr.hp.com>
Rendered body type: text/plain

Phil Karn <karn@qualcomm.com> said:

>So my main concern with the existing rules is that they keep us from
>using modulation and coding methods that conserve RF power. The result
>is a larger average amount of interference power to both narrowband
             ^^^^^^^
>and wideband users. ...

>Yes, it is possible to concoct worst cases where spread spectrum (or
>even a power-efficient wideband modulation scheme) could occasionally
>interfere with a nearby narrowband user. ...

>And this is ultimately why we should encourage spread spectrum on all
>of our bands, especially the most crowded ones.

Sigh. Let me try explaining this once again.

Spread spectrum works fine for something like a cellular telephone system
where the locations and power levels of all stations are well-controlled.
But on the amateur bands, stations are randomly distributed.

There is no need to "concoct worst case" scenarios: Most amateur
communication requires contacting a station that is much farther away
than other nearby stations.

If that nearby interfering station is using wide-band modulation, then the
interference potential is much greater, even though the *average* power
may be less. A decent communications receiver can handle narrowband
interference around 100 dB stronger than the noise level. Spreading the
interfering signal by, say, a factor of 100 only reduces the interference
by 20 dB. Narrowband has something like an 80 dB advantage over
broadband in this regard.

>I believe Robert Glassey made an earlier comment along these lines
>when he said that SS communications assume local cellular-style
>communications. And he was quite right, if by "cellular style" he
>meant heavy geographical reuse of spectrum, with relaying rather than
>high power used to reach more distant stations. "Cellular style" does
>NOT mean you're unable to talk DX, you just may not be able to do it
>directly. (I routinely "work" hundreds of miles with my little 300mw
>handheld 890 MHz unit...)

Cellular-style communications doesn't work on DX bands, because
nearby stations are typically very much weaker than distant stations.
(Because ground wave propagation is much lossier than sky wave.)

Even on the VHF/UHF bands, I have a hard time seeing how it would work.
If you are 100 yards from the nearest station, and 10 miles from the
next nearest (a not untypical situation), then you and your neighbor
will be able to relay between yourselves with no problem, but will have
trouble relaying to the next station in the chain over the +85 dB
interference from down the street. With narrow-band modulation, that
wouldn't be a problem.

>This strikes at another major aspect of ham radio: the
>"self-sufficient ethic". ...

(Much ad-hominum argument against narrow-band modulation omitted.)

>Spread spectrum turns this culture on its head. Not only does the
>average ham have to accept the obviously absurd notion that a wideband
>signal can be more spectrally efficient than a narrowband signal, ...

"Can be" but not "is" in this case. I guess I'm one of those ignorant
average hams...

AL N1AL

12. SS on HF

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Thu, 1 Aug 1996 15:06:04 -0800 (PDT)

To: hfsig@tapr.org
Message-ID: <199608012206.AA232007164@polecat.sr.hp.com>
Rendered body type: text/plain

forrerj@peak.org (Johan Forrer) said:

>2) How many SS particpants does it take to start raising the noise floor to
>a point where it becomes objectionable? 2, 10, 100, 1000 ?   ...

It only takes 1 if he is more than 20 dB or so louder than the signal you
are trying to receive. On HF, it is very rare to be working one of the
loudest signals on the band. Even for normal rag chewing, there are
typically a number of stations on the band that are much stronger than the
station you are talking to. When working a weak DX station, *most* of
the signals on the band are more than 20 dB louder. If even one of those
is running spread spectrum, you're out of luck.

If the SS station is running low power due to some dynamic power control
scheme, then that improves things somewhat. But even if that 20 dB number
becomes 30 or 40 dB, one SS station will still wipe out the band for a
large number of narrowband users. And by the way, other SS users will
have the same problem.

AL N1AL

13. Re: [HFSIG:1409] SS on HF

From: forrerj@peak.org (Johan Forrer.)
Date: Thu, 01 Aug 1996 16:59:06 -0800
To: hfsig@tapr.org
Message-ID: <199608020010.RAA06070@PEAK.ORG>
Rendered body type: text/plain

Hi Al,

>forrerj@peak.org (Johan Forrer) said:
>
>>2) How many SS particpants does it take to start raising the noise floor to
>>a point where it becomes objectionable? 2, 10, 100, 1000 ?   ...
>
>It only takes 1 if he is more than 20 dB or so louder than the signal you
>are trying to receive.
>On HF, it is very rare to be working one of the
>loudest signals on the band. Even for normal rag chewing, there are
>typically a number of stations on the band that are much stronger than the
>station you are talking to. When working a weak DX station, *most* of
>the signals on the band are more than 20 dB louder. If even one of those
>is running spread spectrum, you're out of luck.

To help me understand it a bit better: how much power (EIRP) would that
single SS station need to run to be 20 dB over your DX station?

Assume for argument sake that its a station in another state thats running
DS SS spread over 1 MHz bandwidth thats not skip to you; he's beaming in
your direction and that you receive the DX at S1 level?

--Johan

14. SS Stuff

From: Rick Booth <rick@itron-ca.com>
Date: Thu, 1 Aug 96 18:17:47
To: hfsig@tapr.org
Message-ID: <Chameleon.838949133.rick@rickb.itron-ca.com>
Rendered body type: text/plain

With regard to interference between SS systems and "conventional" users:

Consider a frequency hopper with the following properties. Using 2.4 KHz as the bandwidth of
the radio and assuming a baud rate of 300 (the most allowed by the FCC) and assuming 4 bits per
baud then in order to burst a bit sync preamble plus 10 bytes of data then the signal will have
to dwell on the frequency for about 84 milliseconds (80 bits plus 20 sync bits). For 40 meters
the band is only 300 KHz wide so the band is divided into about 125 slots to hop into. This
means every 10 seconds on the average the SS system will jump into a typical 2.4 KHz single
sideband "slot" and camp there for 84 milliseconds not counting filter thump from the 400
microsecond transient of a 2.4 KHz filter.

This might be annoying if the SS guy is down the street.

Rick
W6NZK

Rick Booth
'95 900SS SP
E-mail: rick@itron-ca.com
----

15. Re: [HFSIG:1411] SS Stuff

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 1 Aug 1996 23:42:11 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608020642.XAA28458@servo.qualcomm.com>
Rendered body type: text/plain

Rick seems to be describing a slow hopper, i.e., a system with
multiple data bits per hop. This produces a fairly long dwell time on
each channel. I was envisioning a "medium speed" hopper, with
something like one symbol per hop. That makes for a much faster hop
rate and a shorter dwell time on each channel.

As an aside, fast hopping is defined as multiple hops per modulation
symbol, and that implies a noncoherent combining loss because it's
difficult or impossible to maintain phase coherence across a
hop. (Even if the radios could do it, chances are you're hopping by
more than the coherence bandwidth of the channel.) Because of this
loss, fast hopping is generally recommended only against active
intelligent jamming, or when the coherence bandwidth of the channel is
unusually narrow, i.e., when the delay spread is long enough to
otherwise cause considerable ISI at the desired modulation symbol
rate.

Phil

16. Re: [HFSIG:1409] SS on HF

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 1 Aug 1996 23:59:10 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608020659.XAA28468@servo.qualcomm.com>
Rendered body type: text/plain

>It only takes 1 if he is more than 20 dB or so louder than the signal you
>are trying to receive. On HF, it is very rare to be working one of the

>becomes 30 or 40 dB, one SS station will still wipe out the band for a
>large number of narrowband users. And by the way, other SS users will
>have the same problem.

Again, this analysis seems to assume direct sequence. Frequency
hopping is quite different thanks to the bursty nature of the
interference.

Phil

17. Re: [HFSIG:1401] Re: Shared SS

From: Frode Weierud <frode@dxcern.cern.ch>
Date: Fri, 2 Aug 1996 10:09:07 +0200 (MET DST)
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <Pine.ULT.3.91.960802091805.15741A-100000@dxcern.cern.ch>
Rendered body type: text/plain

On Thu, 1 Aug 1996, Walt DuBose - K5YFW wrote:

>   Way back there in my AFRes days, we were running SSB on several
>   channels in the 8.4 - 8.9 MHz and in the 11.5 MHz range and the spooks
>   about 300 meters were running SS between 3 -15 MHz to the same
>   locations that we were. Propagation to the location we were both
>   talking to was fair on 8 MHz and poor on 11 Mhz with our equipment.
>   The spooks had very good communications. We were running 500 Watts
>   PEP and they were running less that a 100 watts IAW their specs.
>   Also, it didn't matter if they were on-the-air or not, it had no
>   affect on our communications and I don't recall seeing an
>   increase in channel noise...it darn sure didn't bother the marine
>   CW station on the channel ajacant to our main channel (frequency).
>   Also, there was another spook station located about 5 miles away
>   and we couldn't tell when either was transmitting unless we called
>   them on the fieldfone and asked. I might also mention that they
>   were running broadband antennas and we were running dipoles so our

> antennas were better.
>

As far as I know we already have SS transmissions on most if not all of our HF bands. Some years back I visited a military installation using SS on HF and I asked them if they blanked out the amateur bands and marine bands etc. The answer was NO. Their reply was that as we could not hear them they would not interfer and hence there was no reason not to include the ham bands in their frequency pools. I don't know what power they normally used, but they were equipped with 1KW and 10KW power amplifiers and the antennas were LPAs and disk-cones.

This is a one off experience, but I have the feeling that their attitude concerning SS and interference might be wide-spread and that many military/government organisations using SS will think the same way. Their main reason for using SS is the low probability of intercept and anti-jamming properties.

It would be interesting to hear if anybody else have some experience from military/government use of SS on HF, and specially if anybody is situated close to such an installation they could perhaps let us know if they have ever been able to detect the presence of SS signals.

73's Frode, F/LA2RL

Frode Weierud Phone : +41 22 7674794
CERN, SL, CH-1211 Geneva 23, Fax : +41 22 7679185
Switzerland E-mail : Frode.Weierud@cern.ch

18. SS and the Amateur culture

From: Robert.Glassey@nmp.nokia.com
Date: Fri, 2 Aug 96 12:51:50 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022934c2@MHS>
Rendered body type: text/plain

> I believe Robert Glassey made an earlier comment along these lines
> when he said that SS communications assume local cellular-style
> communications. And he was quite right, if by "cellular style" he
> meant heavy geographical reuse of spectrum, with relaying rather than

[snip] ... Yip, thats what I meant.

> This strikes at another major aspect of ham radio: the
> "self-sufficient ethic". Despite the literal meaning of the third
[snip]

> gain credit. Big antennas and massive linear amplifiers command a lot
> of respect, while resorting to a repeater is at least an admission of
> inferiority, or even downright dishonest (if used in a competition).
[snip]

> Spread spectrum turns this culture on its head. Not only does the
[snip]

> he is also being asked to believe that repeaters can be more
> spectrally efficient than simplex! That is, a network of relatively
[snip]

> up and you can have a network that, overall, is much *more* reliable
> than DX simplex thanks to the shorter inter-node propagation paths.
[snip]

> work DX on HF. You just wouldn't necessarily do it
> directly. Unfortunately, for a lot of hams this wouldn't be the
> same.

There's a few edited clips that I think sum up the angle you seem to be taking on this issue. I've already said my peice here, but I have just a few more comments.

You have quite rightly noted two things, 1. that spread spectrum culture is quite different from traditional amateur culture, and 2. that spread spectrum can make much more efficent *use* of RF resources.

One has got to ask, taking a wide view of amateur radio in general, how important ultimate efficency is, and why amateur culture is the way it is.

The valid arguments based on efficency are more in line with commercial objectives for the use of spectrum. They are more concerned with maximum number of users, level of service, and end user features. If amateur radio is operated according to such commercial objectives, it would make perfect sense for authorities to SELL the spectrum to commercial interests, since they will be able to provide the same type of service, suplimented with other telecoms, and may well be able to provide an even more efficent and reliable service, while making money for the authorities and the telecommunications industry.

There are already bands allocated for these services, and commercial interests are operating on them. Amateur radio has always *justfied*its* *existance*, not as an end user service provider (that all but regulated against), but as an experimental service for the self training of amateurs in the techniquies of radio communications. This is all encompassing, covering a wide variety of modes and applications and is for everyone from the beginner to the expert. One mode cannot take precedence at the expence of all others. Efficency at the expence of the aims and diversity of the service cannot be justified. As a cooperative additional mode, efficency would be welcomed, as was SSB. But SS has a long way to go before it could be accepted as such, if at all.

'You can have any colour you like, so long as its black.' - another expert in efficency and commercialism.

Rob

19. RE: [HFSIG:1410] Re: SS on HF

From: Jon Bloom <jbloom@connix.com>
Date: Fri, 2 Aug 1996 00:46:50 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB804F.370D2280@jbloom.connix.com>
Rendered body type: text/plain

Johan Forrer.[SMTP:forrerj@peak.org] wrote:
>To help me understand it a bit better: how much power (EIRP) would that
>single SS station need to run to be 20 dB over your DX station?
>
>Assume for argument sake that its a station in another state thats running
>DS SS spread over 1 MHz bandwidth thats not skip to you; he's beaming in
>your direction and that you receive the DX at S1 level?

Why would you assume he's in another state? I'd be more interested in the case where he's on the other side of town. Put it another way: that SS station may be in another state from me, but he's on the other side of town from *somebody.* With a decent receiver, I can handle a narrowband signal on the other side of town.

Over the past 20 years or so we hams have beat on the receiver manufacturers to make them design radios with 100+ dB of dynamic range. What that means is that we expect to be able to handle off-channel signals that are 100+ dB above the noise floor. The reason we want that kind of performance is because we *get* signal levels like that entering our receivers. Say there's a narrowband station nearby that I'm receiving at a level just within the 100-dB dynamic range of my receiver, so I'm not experiencing interference from his signal. If he now switches over to SS, he must reduce power and/or spread his signal to reduce the signal received in my receiver within the passband by 100 dB to be at my noise floor. (Of course, I'm willing to cut a few dB of slack.) Is that feasible? That's the question.

By the way, just what HF band do you plan to use for a 1-MHz-wide signal?

-- Jon

20. Re: [HFSIG:1407] Re: Shared SS

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Fri, 2 Aug 1996 09:09:30 -0400
To: hfsig@tapr.org
Message-ID: <201FFA80.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

Are you saying that his COMPUTER is causing the interference? I find
     it hard to believe he is causing that much interference 'down the
     street.'

     73s de
     Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1407] Re: Shared SS
Author: hfsig@tapr.org at BALT.SMTP
Date: 8/1/96 2:59 PM

At 12:50 01/08/96 -0500, you wrote:
> I'd like to know a little more about this 'computer brat down the
> street.' How do you know he is using SS?

He isn't actually. In fact he is a member of Joe Public who has just
turned his P.C on (The grey box with all the wires hanging out the back).

:- }

- Charles

21. Re: [HFSIG:1416] Re: SS on HF

From: forrerj@peak.org (Johan Forrer.)
Date: Fri, 02 Aug 1996 07:40:30 -0800
To: hfsig@tapr.org
Message-ID: <199608021452.HAA06591@PEAK.ORG>
Rendered body type: text/plain

Hi Jon,

>Why would you assume he's in another state? I'd be more interested in the
>case where he's on the other side of town. Put it another way: that SS
>station may be in another state from me, but he's on the other side of
>town from *somebody.* With a decent receiver, I can handle a narrowband
>signal on the other side of town.

I agree that local interference might be very strong. However, my experience
being an avid CW operator (I have a piece of wallpaper saying I won the CQWW
CW single band single op contest once upon a time), is the big signals from
elsewhere, not from local sources. So, thats what I was theorizing about a bit.

>
>Over the past 20 years or so we hams have beat on the receiver manufacturers
>to make them design radios with 100+ dB of dynamic range. What that means
>is that we expect to be able to handle off-channel signals that are 100+ dB
>above the noise floor. The reason we want that kind of performance is because
>we *get* signal levels like that entering our receivers. Say there's a
>narrowband station nearby that I'm receiving at a level just within the
>100-dB dynamic range of my receiver, so I'm not experiencing interference
>from his signal. If he now switches over to SS, he must reduce power and/or
>spread his signal to reduce the signal received in my receiver within the
>passband by 100 dB to be at my noise floor. (Of course, I'm willing to cut
>a few dB of slack.) Is that feasible? That's the question.

Yes, thats basically the gist of what I was after. How much power would it
take from a remote DS SS station to raise the noise floor to 20 dB over a
weak (say S1 level) DX signal. Just curious, because I suspect it would be
enormous (by amateur standards).

>
>By the way, just what HF band do you plan to use for a 1-MHz-wide signal?
>
>-- Jon
>
>

Just a theoretical question to see if I understand the nature of the SS PSD
implications. Perhaps you can give some insight; Would any degree of QSY
have an effect, i.e., how white is the spectrum really if there we have
light SS occupancy?

Thanks,

--Johan

22. Re: [HFSIG:1414] Re: Shared SS

From: Charles Hoffman <choffman@pelican.davlin.net>
Date: Fri, 02 Aug 1996 13:06:23 -0500
To: hfsig@tapr.org
Message-ID: <3202439F.3957@pelican.davlin.net>
Rendered body type: text/plain

Frode Weierud wrote:
>
> On Thu, 1 Aug 1996, Walt DuBose - K5YFW wrote:
>
> > Way back there in my AFRes days, we were running SSB on several
> > channels in the 8.4 - 8.9 MHz and in the 11.5 MHz range and the spooks
> > about 300 meters were running SS between 3 -15 MHz to the same
> > locations that we were.
> > didn't matter if they were on-the-air or not, it had no
> > affect on our communications and I don't recall seeing an
> > increase in channel noise...it darn sure didn't bother the marine
> > CW station on the channel ajacant to our main channel (frequency).
> > Also, there was another spook station located about 5 miles away
> > and we couldn't tell when either was transmitting unless we called
> > them on the fieldfone and asked. I might also mention that they
> > were running broadband antennas and we were running dipoles so our
> > antennas were better.

I have known Walt many years; his military comms experience is more valid here than supposition of what a hypothetical computer guy down the street will do or whatever hypothetical situation imagined. We need to pay attention to Walt's experience reported here.

> >
>
> As far as I know we already have SS transmissions on most if not all of
> our HF bands. .. a military installation using SS
> on HF...asked them if blanked out the amateur bands and marine
> bands etc. The answer was NO... they were equipped with 1KW and 10KW power amplifiers ...antennas were
LPAs

Typical of HF SS mil/gov systems I am familiar with.

> Their main reason for using SS is the low probability of intercept and
> anti-jamming properties.

Exactly the point!

>
> It would be interesting to hear if anybody else have some experience from
> military/government use of SS on HF, and specially if anybody is situated
> close to such an installation they could perhaps let us know if they have
> ever been able to detect the presence of SS signals.

My amateur radio station is situated on a ranch in South Texas eight miles directly off the backside of the recently installed "over the horizion radar" (OHR) built by Raytheon. My best guess is that collectively (multiple amps), feeding 13 phased LPA's run about 200,000 to the antennas (not erp). Frequency range, from observing the two banks of LPAs appears to cover from 4 mHz to 30 mHz. The system apparantly uses various modulation methods but the mode is SS, and it does cover the complete spectrum including amateur radio frequencies.

Considering the power, the nature of the LPA F/B characteristics and the distance to the antenna, it is significant that I did not have any harmful interference, including noticable receiver frontend sensitivity reduction. During their testing phase I sometimes could tell when they were on the air. Sometimes, and barely discernable. Later under normal operations I never hear them.

By comparison, from the same site they have two propogation measurement devices. One is an oblique chirp sounder which squirts away from me and the other is a F1F2 height sweep sounder using a NVIS dipole. The later runs 10KW, and sweaps the spectrum of interest with what sounds like a A1 carrier. It sweeps through my receiver passband in no more than 400ms but eats my lunch. The receiver is useless for three or four seconds. The s-meter always hits the far peg with a "clink"!

The LPA-backside SS signal is considerably greater rf field at my QTH than the sounder signal but is unnoticed. I have heard it however; one of the modes sounds like tinkling ice cubes in a pitcher of iced tea, as its being poured. I do believe that is what killed the grass at the ranch, not the drought.

Hope this is of some interest. 73 Rik K5SBU

>
> 73's Frode, F/LA2RL
>
> Frode Weierud Phone : +41 22 7674794
> CERN, SL, CH-1211 Geneva 23, Fax : +41 22 7679185
> Switzerland E-mail : Frode.Weierud@cern.ch

23. SS on HF

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Fri, 2 Aug 1996 13:00:07 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <199608022000.AA220946008@polecat.sr.hp.com>
Rendered body type: text/plain

forrerj@peak.org (Johan Forrer.) said:

>>>2) How many SS particpants does it take to start raising the noise floor to
>>>a point where it becomes objectionable? 2, 10, 100, 1000 ? ...
>>
>>It only takes 1 if he is more than 20 dB or so louder than the signal you
>>are trying to receive.
>
>To help me understand it a bit better: how much power (EIRP) would that
>single SS station need to run to be 20 dB over your DX station?
>
>Assume for argument sake that its a station in another state thats running
>DS SS spread over 1 MHz bandwidth thats not skip to you; he's beaming in
>your direction and that you receive the DX at S1 level?

The propagation loss varies widely with different paths. At any given time, signals on a DX band typically vary from the noise level (less than a tenth of a microvolt, around -130 dBm) up to the millivolt range, -30 dBm or so. (This assumes skywave propagation. It can be worse if there is another ham station in your neighborhood.) A good communications receiver's blocking level might be around 0 dBm, so it can handle typical band conditions with no problems. But if one of those -30 dBm signals has spread its signal over a 300 kHz band, it will be putting -50 dBm interference into each 3 kHz narrow-band channel.

You would have to reduce the power by something on the order of 80 dB in order not to cause interference. 80 dB below a legal-limit 1500W signal is 15 microwatts, much too low for reasonable communications capability on HF. You can argue with my numbers, but you would have to find a very large error in them in order to invalidate my argument.

If the spread spectrum signal is a frequency-hopping type, then it will be putting a full-strength (non-spread) signal into each channel at some repetitive rate. In order to cause no interference at all, the power must be reduced even 20 dB more than for the example above. True, frequency hopping does reduce the percentage of time that the interference occurs, but that percentage increases the more SS stations there are operating on the band at any given time.

AL N1AL

24. Re: [HFSIG:1400] Shared SS

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Fri, 2 Aug 1996 15:46:39 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608022246.PAA05002@unix.ka9q.ampr.org>
Rendered body type: text/plain

>Narrowband users already dynamically share spectrum. They QSY to avoid QRM.

That's not dynamic enough. You're still dedicating a chunk of spectrum to a particular QSO for as long as it lasts. It goes to waste whenever neither party is talking, and even during speech pauses during a transmission.

The kind of dynamic sharing I'm talking about exploits the peak/average ratios of voice conversation, not to mention the enormously larger ratios present in most computer commnications that is the rationale for packet switching. It puts all the users and all the spectrum in one big pool to exploit the law of large numbers.

Qualcomm CDMA does exactly this with its variable rate vocoder. When you talk, the data rate kicks up to 9.6 kb/s (or 14.4 kb/s in newer systems). When you stop talking, it drops back to 1.2 kb/s. The average data rate is something like 40% of the peak in a typical conversation.

A typical CDMA sector handles 25-35 calls in each 1.25 MHz RF channel, and that's enough to gain considerable benefit from pooling. Only if everyone talks at exactly the same moment do you hit the limit, and the probability of that goes down as you make the pool larger.

Again, I won't give you a guarantee that SS will *never* cause interference. I can't, though I can certainly improve the odds with power control, efficient coding and adaptive hopping patterns. But it's just unrealistic to hold SS to such a high standard when

narrowband operations are not. QRM has always been a way of life on
the shared HF bands, and the best we can do is to promote methods that
reduce it on average.

Phil

25. Re: [HFSIG:1420] SS on HF

From: forrerj@peak.org (Johan Forrer.)
Date: Fri, 02 Aug 1996 16:26:23 -0800
To: hfsig@tapr.org
Message-ID: <199608022338.QAA01803@PEAK.ORG>
Rendered body type: text/plain

Alan,

> A good communications receiver's blocking
>level might be around 0 dBm, so it can handle typical band conditions with
>no problems. But if one of those -30 dBm signals has spread its signal
>over a 300 kHz band, it will be putting -50 dBm interference into each
>3 kHz narrow-band channel.
>
>You would have to reduce the power by something on the order of 80 dB in
>order not to cause interference. 80 dB below a legal-limit 1500W signal is
>15 microwatts, much too low for reasonable communications capability on HF.
>You can argue with my numbers, but you would have to find a very large
>error in them in order to invalidate my argument.
>

This is exactly what I was trying to grasp. Thanks for the explanation.

--Johan

26. HF SS Demonstration

From: k6sti@n2.net (Brian Beezley)
Date: Fri, 2 Aug 1996 23:26:18 -0700
To: hfsig@tapr.org
Message-ID: <199608030626.XAA06402@ravel.n2.net>
Rendered body type: text/plain

At this point it seems like everyone has pretty much spoken his piece
regarding the feasibility of mixing spread spectrum and existing HF
narrowband modes.

I'd like to take the discussion one step further by proposing a simulated HF
SS demonstration. The demonstration will give anyone with an HF receiver a
chance to hear what frequency-hopped signals sound like and to judge their
effect on narrowband users.

Here's what I propose: I'll write PC software that hops an HF transceiver
around in the 20-meter band. We'll run the transceiver in USB mode and
drive it with a controlled audio waveform generated by a sound card. We'll
pick a contest-free Saturday and for a 15-minute period, a dozen or more
stations across the country simultaneously will transmit simulated SS.
Interested parties can listen to the band during this demonstration and make
their own judgment about the level of interference generated. You can
monitor conversations in progress and listen for comments. You can record
what you hear. If you're near one the simulated SS stations, you can judge
the level of near-transmitter QRM; otherwise, you'll have to be content to
listen to the skywave signals.

I'll leave it to the SS proponents to specify a power level and devise a
waveform and hopping sequence. I'd suggest a maximum of 50 watts RMS output
power to avoid overheating barefoot exciters. I'd suggest using a noiselike
waveform rather than one with spectral lines to make it audibly less
intrusive. For this demo I'm not volunteering to program an adaptive hop
algorithm--to do it right would take more time and effort than I'm willing
to expend on this experiment.

To simplify things, I'll write the code to control the frequency of a single
brand of transceiver. Unfortunately, I don't own a transceiver that can be
computer controlled, but I do have access to several locally for testing.

There are a couple of nontechnical issues involved in a test like this. The
first is the propriety of intentionally emitting power on the frequency of a
QSO in progress without permission or warning. The second concerns the
legality of such a test. I expect these issues to be addressed in
discussion that follows by those concerned. Personally, I don't have a
problem with either as long as the test is brief. Presumably, none of the

stations volunteering to transmit SS will either.

I don't propose to publicize this demonstration ahead of time except to
announce the time and date on several Internet reflectors a day in advance.

If the SS proponents aren't interested in this demonstration or if I can't
scrounge up enough volunteers, I'll drop the idea. Otherwise, who's game?

Brian Beezley, K6STI
k6sti@n2.net

27. Re: [HFSIG:1401] Re: Shared SS

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sun, 4 Aug 1996 14:49:40 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608042149.OAA00779@unix.ka9q.ampr.org>
Rendered body type: text/plain

Way back there in my AFRes days, we were running SSB on several
        channels in the 8.4 - 8.9 MHz and in the 11.5 MHz range and the spooks
        about 300 meters were running SS between 3 -15 MHz to the same
        locations that we were. Propagation to the location we were both

To be perfectly fair to the "other side", I have to point out that the
process gain (spreading range) here was much larger than a single ham
band. If frequency hopping were used, I'd expect them to be virtually
undetectable except on a wideband spectrum analyzer close to their
transmitter.

Phil

28. Re: [HFSIG:1424] Re: Shared SS

From: Charles Hoffman <choffman@pelican.davlin.net>
Date: Sun, 04 Aug 1996 18:18:35 -0500
To: hfsig@tapr.org
Message-ID: <32052FCB.4A9A@pelican.davlin.net>
Rendered body type: text/plain

Phil Karn wrote:
>
>   Way back there in my AFRes days, we were running SSB on several
>   channels in the 8.4 - 8.9 MHz and in the 11.5 MHz range and the spooks
>   about 300 meters were running SS between 3 -15 MHz to the same
>   locations that we were. Propagation to the location we were both
>
> To be perfectly fair to the "other side", I have to point out that the
> process gain (spreading range) here was much larger than a single ham
> band. If frequency hopping were used, I'd expect them to be virtually
> undetectable except on a wideband spectrum analyzer close to their
> transmitter.
>
> Phil

        Interesting, but who's watching? Just be invisible like everyone else
on SS! Ho ho. How many amateur radio stations are on the ham bands on
SS? How many non amateur stations are on the ham bands on SS? Does it
matter? Who is minding the store?

        73 Rik K5SBU

29. Re: [HFSIG:1424] Re: Shared SS

From: fperkins@onramp.net
Date: Sun, 4 Aug 1996 18:33:16 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608042333.SAA13962@mailhost.onramp.net>
Rendered body type: text/plain

Hi Phil and the HF Gang,

I looked around a bit this weekend trying to find a basic discussion
on the channel capacity of "random" packet transmissions. I am trying
to gain some insight into the "18% barrier".

Could you suggest a reference?

73 Frank WB5IPM

30. Re: [HFSIG:1402] Wide band modes

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sun, 4 Aug 1996 17:34:40 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608050034.RAA00887@unix.ka9q.ampr.org>
Rendered body type: text/plain

Robert, thanks for your long and thoughtful note. There was so much in there, I just finished digesting it all.

>It is indeed. I have no arguments here. I've long suspected as much, >although I've never had the numbers that show it. Although even very >simple ECC can make back most of this 35-40db difference. ie and AMTOR

Quite true. To paraphrase your local real estate agent, three things are most important in dealing with Rayleigh fading: diversity, diversity and diversity. Even simple repetition coding (the simplest form of ECC) is a significant benefit. But properly constructed codes use the redundancy more efficiently, and work better as a result.

>I guess without any extra power, the signal would be above the threshold >50% of the time. Thus we have a 50% BER. A 1/2 rate ECC would ideally >deal with this, although I guess a bit more redundancy would be usefull.

A 50% BER corresponds to zero capacity, which is the worst any communication system can do. (If the BER were greater than 50%, you could simply invert it to make the BER less than 50%, and you'd have some non-zero capacity).

Since the capacity of a BSC (binary symmetric channel) with a 50% BER is zero, no code can possibly fix this. However, a BEC (binary erasure channel, i.e., a BSC supplemented with a third "erasure" output) is a different story. The capacity of a BEC, assuming no "hard" errors, is as you expect very close to 1 minus the erasure rate. A rate 1/2 ECC would work pretty well here assuming proper attention is given to interleaving.

>In the light of this 15dB fade margin, I wonder how much effect soft >decoding would really have. Would erasure decoding be just as good? >What I'm geting at is PACTOR II's K=9 or is it 11 contraint length. >Would k=5 be just as good? Or could we just go for plain Reed Soloman >coding? This would make PC implementation of an HFSIG protocol more >realistic.

This is a very interesting question. You can think of erasure decoding as a form of "softened" decision decoding where the received symbols can take on three levels (zero, one and "don't know"), and it's certainly true that a little bit of "softness" goes a long way. It does make Reed-Solomon decoders look more interesting since they can handle erasures fairly easily but not full-blown soft decisions.

But I see nothing wrong with Viterbi decoding per se, and there is probably still some gain to be had from soft decision decoding. I think I have some references that can quantify this. Sure, Viterbi is slower than Reed-Solomon but with all the cycles we have these days it's just not a major factor at the data rates we're likely to use. (My K=7 decoder runs at 155 kb/s on a P90).

>But getting back to your spreading argument, I've just suggested >trippling the bandwidth to overcome fading, or have I?

Exactly!! FEC inherently "spreads" bandwidth, and that makes it as much a "spread spectrum" scheme as frequency hopping or direct sequence chipping. It certainly does cause you to run into the FCC emission bandwidth limits, which are the basic problem here.

>I recall Ungerboch (I can never spell that) said that the solution was >to go for multi levels, even if you cannot actually resolve each one

I'm not surprised Ungerboch said this. He invented trellis coding, and it's been extremely successful on bandwidth limited channels (e.g., telephone modems). But beyond the inappropriate use of the bandwidth-limited model for the power-limited radio channel, fading introduces some serious practical problems. If the symbols have unequal energy, the demodulator thresholds would depend on signal amplitude, and this changes rapidly and unpredictably with fading. This is in fact one of the most serious drawbacks of on-off-keyed CW. Ever tried to copy it on a channel that's fading at about the same rate as a CW symbol element?

In my opinion, all these higher-order QAM schemes in Clover et al are
going down the wrong path. They certainly help account for their
extremely high Eb/N0 requirements when operating at high data
rates. It would be much better to let them relax their bandwidth
constraints and just scale up their more power-efficient schemes like
BPSK to wider channels.

>Not inevitible, but I can still see that a wider band signal will have
>faster fades that are more easilly dealt with. I think the wider
>bandwidth helps when you consider interference from other users. Still a
>narrow a bandwidth allows more users in the same wide bandwidth without
>interference. 6 of one, half dozon of another as they say. I dont think
>either can make full use of the bandwidth.

It's not that the fading is faster in the wideband case, it's just
another way to get more diversity.

You can get diversity in different ways: time, frequency, space,
polarization, etc. All that matters is that the diversity is over
sufficient distance to get some independence in the fading. That is,
you need to diversify the effect of any given user data bit over
considerably more than one coherence time, one coherence bandwidth,
etc. It's sufficient to diversify over just one dimension, but you'll
need more of it than if you use several. For example, real-time voice
may have some serious delay constraints that limit the use of time
diversity, so frequency diversity is perhaps even more useful than it
would be in a non-real-time packet data system where you could afford
to diversify over many seconds of time.

I disagree strongly with the "6 of one, half dozen of the other"
characterization. It is very clearly the case that what you spend in
bandwidth is more than bought back in extra interference resistance.
This is most clearly the case when the extra bandwidth allows a decrease
in Eb/N0 through coding, but there are other practical benefits from
spreading even when the AWGN Eb/N0 is unaffected (as it is with direct
sequence or frequency hopping): the increased "trunking" or "pooling"
efficiencies of a large group of users dynamically sharing a large pool
of spectrum, and the benefits of frequency diversity that I've already
mentioned.

>Hmm, this is interesting, expecially the 5dB increase over the BPSK
>case. I wonder what it would be over the 64-QAM case.

The literature I have doesn't give any results for QAM since it's
considered in appropriate for the fading channel. Even coherent BPSK
is inappropriate because of the difficulty in tracking carrier phase
(differentially detected BPSK is recommended instead.)

>How resistant would this M-ARY scheme be to other signals? I wonder if

Depends on the nature of the QRM. Continuous unmodulated carriers
could be handled rather easily with adaptive notch filters. But strong
intermittent QRM would be harder to deal with. That's why you really
want to combine this with frequency hopping; by using only a small
subset of all the available frequencies at any given time, you reduce
the chances of landing on strong QRM to where coding can easily handle it.

>I'm actually fairly well convinced that with a properly designed digital
>radio, with a wide bandwidth and demod controlled AGC could overcome the
>near far problem quite well, especially if spread enough so that local
>impulses cause a BER of < 10%. I even like the idea!

Great!

>The concern is about sharing the band with other users. I dont think

I agree, and actually that's why I would support dividing up the band
into "DX" and "regional" subbands to mitigate the near-far problem.
Each subband would still allow wideband modulation within it, but only
for the right purpose (i.e., DX or local). I think this would go a
long way towards solving the problem, don't you?

Phil

31. Re: [HFSIG:1406] Re: Advantages of SS

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sun, 4 Aug 1996 22:00:45 -0700 (PDT)

To: hfsig@tapr.org
Message-ID: <199608050500.WAA01008@unix.ka9q.ampr.org>
Rendered body type: text/plain

>1) If assumed that the center frequency of different DS SS operations are
>randomly chosen, what would that do to the near-far problem, i.e., if you
>use DS SS and experiencing interference from your neighbor DS SS operator,
>why not QSY out of harm's way of the offending signal envelope? This assumes
>that contacts are initially established on some known frequency. Am I right
>in assuming that if only two parties are involved in DS SS, that the PSD
>does have a maxima somewhere, even though it is a wideband, i.e, not quite
>white over the full bandwidth?

Johan,

You assume that there's little if any filtering after the spreading
operation. Most real systems I know do in fact filter, both to
flatten the in-band spectrum and to knock off the sin(x)/x
sidelobes. Qualcomm CDMA uses a particularly sharp FIR filter because
of cellular carrier requirements. To get the sharp skirts we let the
passband ripple somewhat. The resulting display on a spectrum analyzer
closely resembles the top of Bart Simpson's head, so around here we
often call each composite CDMA waveform a "Bart's head".

>2) How many SS particpants does it take to start raising the noise floor to
>a point where it becomes objectionable? 2, 10, 100, 1000 ? If the number is
>1000, would it be realistic, given a comparable narrow-band situation, to
>expect anyone to be able to conduct a QSO with that occupancy? Like a
>contest with 1000 RTTY operators calling simultaneously in the 14.090 -
>14.065 20 meter segment.

This depends on the process gain, required Eb/N0s and relative signal
strengths. The math is actually pretty straightforward.

Consider a DS/SS system with coding and spreading such that an Eb/N0
of 6 dB is required and the ratio of RF bandwidth to user data rate is
128:1. (These are the approximate parameters of Qualcomm CDMA).

The process gain is 10*log10(128) ~= 21 dB. For a required Eb/N0 of 6 dB,
the overall SNR required is 6 - 21 = -15 dB. This figure is sometimes
referred to as Ec/I0, the ratio of the energy per chip to the interference
spectral density.

If everyone's power is matched at the receiver, you can tolerate as
many as 31.6 other users before their aggregate interference brings
the Eb/N0 down to 6 dB. Note this is less than the 128 users you could
theoretically support by just dividing up the total bandwidth into 128
equal size pieces. But that assumes no thermal noise, i.e., that it
would be easy to operate at the higher Eb/N0 that's required without
bandwidth-expanding coding, especially on a fading channel. It also
ignores any savings from better dynamic sharing, and it ignores the
benefits from being able to geographically reuse frequencies more
densely because of the far greater resistance to co-channel
interference.

Phil

32. Re: [HFSIG:1408] SS on HF bands

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sun, 4 Aug 1996 22:26:25 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608050526.WAA01021@unix.ka9q.ampr.org>
Rendered body type: text/plain

>Spread spectrum works fine for something like a cellular telephone system
>where the locations and power levels of all stations are well-controlled.
>But on the amateur bands, stations are randomly distributed.

Most cellular users are randomly distributed too. But they do generally
talk only to the nearest base station.

A somewhat better model is the DARPA SURAN (Survivable Radio Network)
project of the 1970s and 1980s. It assumed randomly distributed packet
radios operating in a simplex fashion. It used spread spectrum.

>If that nearby interfering station is using wide-band modulation, then the
>interference potential is much greater, even though the *average* power
>may be less. A decent communications receiver can handle narrowband
>interference around 100 dB stronger than the noise level. Spreading the
>interfering signal by, say, a factor of 100 only reduces the interference

>by 20 dB. Narrowband has something like an 80 dB advantage over
>broadband in this regard.

This is true as far as it goes, but 100 dB adjacent channel rejection
seems a little generous. 60 dB is a more reasonable estimate, I think.
Whatever.

>Cellular-style communications doesn't work on DX bands, because
>nearby stations are typically very much weaker than distant stations.
>(Because ground wave propagation is much lossier than sky wave.)

Ah, but wouldn't this tend to mitigate the near-far problem?

>(Much ad-hominum argument against narrow-band modulation omitted.)

I don't see why you have to label this "ad-hominum". I think it's a
fair characterization of the "ham ethic" as I see it.

I guess it boils down to what you think ham radio is all
about. Reliable communications? That description hardly fits direct
long-haul HF. Easily accessible to those without large lots, liberal
housing covenants and money for big antennas and power amplifiers? I
think not. The popularity of VHF/FM and repeater networks shows there
are a lot of hams who are interested primarily in reliable local
communications with small stations, and who aren't purists when it
comes to using relays. Spread spectrum promises to provide the same
capabilities over wider areas on HF.

I confess that perhaps my view of ham radio is somewhat different than
yours. About the only activity that has a hope of justifying our bands
for very long is technical self-training, experimentation and
advancement. That means keeping up with the state of the art in the
non-amateur world, and spread spectrum is certainly one technology
where we trail the commercial and military world. While I don't really
have anything against DXing and contesting, I really don't see how
they help justify ham radio's existence to the outside world; they're
really little more than technological stamp-collecting.

I don't want to do away with traditional hamming, but I don't want it
to choke experimentation either. While I can't guarantee that there
won't ever be any SS interference to narrowband modes, I think it can
be minimized by careful system design and operational coordination.
And whatever risks there are will be worth the benefits. After all,
ours is not a critical safety-of-life service like aviation; we're
supposed to be experimenters, and experiments can fail. That's part of
the process.

Phil

33. Re: [HFSIG:1415] SS and the Amateur culture

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sun, 4 Aug 1996 23:36:58 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608050636.XAA01066@unix.ka9q.ampr.org>
Rendered body type: text/plain

>One has got to ask, taking a wide view of amateur radio in general, how
>important ultimate efficency is, and why amateur culture is the way it
>is.

A very good question.

>There are already bands allocated for these services, and commercial

>interests are operating on them. Amateur radio has always *justfied*its*
>*existance*, not as an end user service provider (that all but regulated
>against), but as an experimental service for the self training of
               ^^^^^^^^^^^^^^^^^^^^^^^^^^^^

BINGO!!

How much serious, state-of-the-art experimentation is going on right now
in amateur radio? You mention single sideband, but that was a long time
ago. As the saying goes, what has ham radio done for the world lately?

Indeed, ham radio doesn't exist in a vacuum. It is not just a hobby.
It is given spectrum that could otherwise be sold for big bucks
because it supposedly gives something back to society as a whole. Now
that emergency communications and international good will have largely

been taken over by cellular telephones and the Internet, respectively, what's left? Technical experimentation and self-training! How much of that really goes on in ham radio these days? Not much. Even worse, the rules are set up to discourage it. That's what I want to change.

Phil

34. Re: [HFSIG:1426] Re: Shared SS

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 5 Aug 1996 00:36:30 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608050736.AAA05953@servo.qualcomm.com>
Rendered body type: text/plain

>on the channel capacity of "random" packet transmissions. I am trying
>to gain some insight into the "18% barrier".

That's the original Aloha result. Keep in mind the conditions: a single narrowband channel; two or more transmissions always result in a collision, i.e., there is no capture effect; there is no channel sensing, so stations transmit whenever they want.

Spread spectrum has entirely different results because collisions are "soft". That is, you can have many stations transmitting at once. At some point the background interference will be too great to get through, but that happens at a load much greater than 2 active transmitters. And again the law of large numbers kicks in, so you can get much closer to the carrying capacity of the channel than you could on a single narrowband (no capture) channel.

Phil

35. Re: [HFSIG:1402] Wide band modes

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Mon, 5 Aug 1996 00:59:48 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608050759.AAA01117@unix.ka9q.ampr.org>
Rendered body type: text/plain

>How resistant would this M-ARY scheme be to other signals? I wonder if
>different convolution codes could be used to provide further resistance
>between signals (getting very spread spectrum like now), with some
>sequential soft decoding being used to distinguish between the different
>stations, rather than just using a majority vote detector. Could this
>provide enough resistance to other stations on the same channel? It may
>be possible to decode 100% a weaker station while a much stronger
>station, or two use the same channel. MARY FSK seems to have quite an
>advantage here.

I meant to comment on the latter part of this paragraph.

You are exactly right -- different convolutional codes can indeed be used to differentiate between signals; it's an entirely viable form of CDMA. If the code rate is low enough, then this is virtually identical to coded direct sequence spread spectrum! In fact, Andrew Viterbi invented such a scheme a few years ago and published a paper about it.

I'm not sure it is of real benefit, though, since you can do almost as well by using a common high rate code (e.g., rate 1/3 or 1/2) and then chipping the encoded symbols with different PN codes in a more conventional direct sequence system.

Although the lower rate code of Viterbi's scheme would give somewhat greater coding gain, most of the gain in a code comes in the first factor of 1/2-1/4 or so. Ultra-low code rates with distinct polynomials for each encoded symbol doesn't gain enough to be worthwhile yet from a complexity standpoint over simply repeating the symbols from a relatively high rate coder -- and that's what the PN chipping effectively does.

Again, the name of the game on a fading channel is *diversity*. Coding and interleaving gives you diversity in the time domain, assuming the interleaving span is much greater than the coherence time. It can also give you some diversity in the frequency domain if the code rate is low enough to increase the signal bandwidth significantly beyond the coherence bandwidth. But just as simple repetition is a viable (if not ideal) form of diversity in the time domain, so it is in the frequency domain -- and that's what direct sequence chipping or frequency hopping both do.

BTW, the original wideband system that John Costas analyzed in his
1959 paper "Poisson, Shannon and the Radio Amateur" was essentially
simple "frequency repetition". That is, he divided up his RF energy
across N identical copies in frequency of the analog waveform and
recombined them on the receiving end. Even with this relatively weak
form of coding (a scheme he described as "intuitively ridiculous"), he
found noticeable improvements in band capacity.

Phil

36. Military SS systems

From: Robert.Glassey@nmp.nokia.com
Date: Mon, 5 Aug 96 11:52:51 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022b01a5@MHS>
Rendered body type: text/plain

On Thu, 1 Aug 1996, Walt DuBose - K5YFW wrote:

> Way back there in my AFRes days, we were running SSB on several
> channels in the 8.4 - 8.9 MHz and in the 11.5 MHz range and the
> spooks about 300 meters were running SS between 3 -15 MHz to the same
> locations that we were. didn't matter if they were on-the-air or not,
> it had no affect on our communications and I don't recall seeing an
> increase in channel noise...it darn sure didn't bother the marine CW
> station on the channel ajacant to our main channel (frequency). Also,
> there was another spook station located about 5 miles away and we
> couldn't tell when either was transmitting unless we called them on
> the fieldfone and asked. I might also mention that they were running
> broadband antennas and we were running dipoles so our antennas were
> better.

Rik K5SBU wrote:

> My amateur radio station is situated on a ranch in South Texas eight
> miles directly off the backside of the recently installed "over the
> horizion radar" (OHR) built by Raytheon. My best guess is that
> collectively (multiple amps), feeding 13 phased LPA's run about
> 200,000 to the antennas (not erp). Frequency range, from observing
> the two banks of LPAs appears to cover from 4 mHz to 30 mHz. The
> system apparantly uses various modulation methods but the mode is SS,
> and it does cover the complete spectrum including amateur radio
> frequencies.
> Considering the power, the nature of the LPA F/B characteristics and
> the distance to the antenna, it is significant that I did not have any
> harmful interference, including noticable receiver frontend
> sensitivity reduction.

These are very interesting results. Is you station behind the LPAs? Do
they ever point in your direction? Have you attempted (out of interest)
to measure interference in front the the LPA's say at a mile or two? Do
the LPAs point away from built up areas?

Over the weekend I did a series of tests on 18MHz with low power pulsed
transmissions to determine their range and interference potential. I'll
post a report soon, however I was able to hear 10mW 2ms carrier bursts
at a distance of 1 mile, and 1W 2ms bursts at 3 miles, using a handheld
scanning receiver with a 70cm long whip antenna!

So how do the military do it? Are they really using kilowatts of power?
How short are their bursts?

It would be very interesting to get a wide band receiver (4-30MHz) with
a fast envelope detector, and very fast peak hold and bring it close to
the SS instalation in Texas. Surely this would make the TX power
detectable, allowing comparisons with other TX sources, and giving an
idea of how the signal is spread.

I expect the military have spent millions on reducing interference (Low
probability of detection) and they use extreemly wide bandwidths. A
radar may well have quite different characteristics compared to a
communications system.

I expect most of their techniques are classified, but there must be
something we can learn from them. From my tests a simple hopping 2ms
burst is unacceptable. How do they do it??

My tests suggest that a DSSS system with 10 microwatts power in 2.4kHz

(100mW in 24MHz) would be detectable at 3 miles. A 10mW carrier 2.4kHz
SNR is about 0dB (still audible above the noise) at 3 miles using a 70cm
long unmatched whip antenna - shown to be at least 30dB less sensitive
than a reasonable amateur station in the presence of ambient noise.

Cheers,

Rob, G0VTQ

37. [HFSIG:1423] HF SS Demonstration

From: Robert.Glassey@nmp.nokia.com
Date: Mon, 5 Aug 96 12:07:09 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022b01b2@MHS>
Rendered body type: text/plain

Hi Brian.

> I'd like to take the discussion one step further by proposing a
> simulated HF SS demonstration.

Aha! exactly what I was thinking. I did just this last weekend. I'll
post a report soon.

> Here's what I propose: I'll write PC software that hops an HF
> transceiver around in the 20-meter band. We'll run the transceiver in
> USB mode and drive it with a controlled audio waveform generated by a
> sound card.

We think alike! This is almost exactly how I did it! except I only used
one SSB channel and transmitted no output when the frequency should have
been in some other channel.

> We'll pick a contest-free Saturday and for a 15-minute
> period, a dozen or more stations across the country simultaneously
> will transmit simulated SS.

.. and I only had one TX and measured interference at a range of 1-5
miles with a hand held scanner.

> I'll leave it to the SS proponents to specify a power level and devise
> a waveform and hopping sequence. I'd suggest a maximum of 50 watts
> RMS output power to avoid overheating barefoot exciters. I'd suggest
> using a noiselike waveform rather than one with spectral lines to make
> it audibly less intrusive.

Tried this, but this form of partical spread does not appear to make the
sound any more acceptible, or more dificult to hear. My tests were 10mW
to 1 Watt, and resulted in significant interference.

Rob

38. Re: [HFSIG:1434] HF SS Demonstration

From: Mike Murphree <n4cnw@pig.net>
Date: Mon, 05 Aug 1996 09:01:19 -0400
To: hfsig@tapr.org
Message-ID: <3205F09F.33C0@pig.net>
Rendered body type: text/plain

Robert.Glassey@nmp.nokia.com wrote:

> > Here's what I propose: I'll write PC software that hops an HF
> > transceiver around in the 20-meter band. We'll run the transceiver in
> > USB mode and drive it with a controlled audio waveform generated by a
> > sound card.
>
> We think alike! This is almost exactly how I did it! except I only used
> one SSB channel and transmitted no output when the frequency should have
> been in some other channel.
>
> . and I only had one TX and measured interference at a range of 1-5
> miles with a hand held scanner.
> Tried this, but this form of partical spread does not appear to make the
> sound any more acceptible, or more dificult to hear. My tests were 10mW
> to 1 Watt, and resulted in significant interference.

This sounds pretty slow.. What was the dwell time on each frequency?
What kind of transceiver was used? DDS?

What modulation was used?

Mike

39. Re: [HFSIG:1434] HF SS Demonstration

From: k6sti@n2.net (Brian Beezley)
Date: Mon, 5 Aug 1996 06:23:28 -0700
To: hfsig@tapr.org
Message-ID: <199608051323.GAA04174@ravel.n2.net>
Rendered body type: text/plain

>> I'd suggest using a noiselike waveform rather than one with spectral lines >> to make it audibly less intrusive.
>
>Tried this, but this form of partical spread does not appear to make the
>sound any more acceptible, or more dificult to hear. My tests were 10mW
>to 1 Watt, and resulted in significant interference.
>
>Rob

Odd. I find stations tuning up--even weak heterodynes--a lot more annoying than SSB splatter or the occasional solar noise burst. The San Diego county noise ordinance lowers the one-hour average SPL noise limit by 5 dB if the noise contains a steady, audible tone "such as a whine, screech, or hum."

Brian Beezley, K6STI
k6sti@n2.net

40. Re: [HFSIG:1433] Military SS systems

From: Charles Hoffman <choffman@pelican.davlin.net>
Date: Mon, 05 Aug 1996 09:17:16 -0500
To: hfsig@tapr.org
Message-ID: <3206026C.111F@pelican.davlin.net>
Rendered body type: text/plain

Robert.Glassey@nmp.nokia.com wrote:
>

> Rik K5SBU wrote:
>
> > My amateur radio station is situated on a ranch in South Texas eight
> > miles directly off the backside of the recently installed "over the
> > horizion radar" (OHR) built by Raytheon.

> These are very interesting results. Is you station behind the LPAs?

Yes, eight miles.

> Do they ever point in your direction?

No, the array is composed of a total of 26 seperate LPA's in vertical configuration, each suspended between towers, and parallel with the next one. The entire antenna array looks from a distance to be about 50 meters by 200 meters perhaps. (A major highway passes about a mile from it) and that is placed on an enormous ground screen observed during installation. The point is that it is not movable. It is steered, according to Raytheon, by phasing to the various individual LPA elements in the array. No moving parts, that is.

> Have you attempted (out of interest) to measure interference in front the the LPA's say at a mile or two?

No I havent, but it could be done from that highway mentioned above, which passes right in front of the array a mile or so in front of it. It is interesting to note, that the newspaper indicated that this OHR system was to be used to observe the caribean area. A line drawn on a map from the array to the Caribean passes directly over the FCC Primary Monitoring Station (KI) eight miles away. The monitoring station recently closed, but for unrelated reasons.
If you are interested, I do have a recording of one of the transmissions from the OHR which I made while Raytheon was testing it. Should anyone be interested it could be made into a wave file and uploaded. I posted inquiries to the Brit's radio newsgroup wondering if their signals could be heard on the ham bands there but no one commented.
In conclusion, it should be noted that aside from likely computer

controllers of this system, it is not particularly new technology. OHR
has been around awhile. Considering that, it could be that someone at
Raytheon, the designer and builder, might be willing to contribute
something of interest to this group.

> Do the LPAs point away from built up areas?

        Aside from the highway traffic passing by in front of the array there
are no built up areas in the immediate vicinity. It is wide open Texas
ranching country. I would say that all of the signal passes over the
King Ranch pastures.
>
> So how do the military do it? Are they really using kilowatts of power? How short are their bursts?

        I doubt if those questions would be answered, but considerable power is
available with the typical Harris transmitters. Yes kilowatts.
>
> It would be very interesting to get a wide band receiver (4-30MHz) with
> a fast envelope detector, and very fast peak hold and bring it close to
> the SS instalation in Texas. Surely this would make the TX power
> detectable, allowing comparisons with other TX sources, and giving an
> idea of how the signal is spread.
>
> I expect the military have spent millions on reducing interference (Low
> probability of detection) and they use extreemly wide bandwidths. A
> radar may well have quite different characteristics compared to a
> communications system.
>
> I expect most of their techniques are classified, but there must be
> something we can learn from them. From my tests a simple hopping 2ms
> burst is unacceptable. How do they do it??
>
> My tests suggest that a DSSS system with 10 microwatts power in 2.4kHz
> (100mW in 24MHz) would be detectable at 3 miles. A 10mW carrier 2.4kHz
> SNR is about 0dB (still audible above the noise) at 3 miles using a 70cm
> long unmatched whip antenna - shown to be at least 30dB less sensitive
> than a reasonable amateur station in the presence of ambient noise.
>
> Cheers,
>
> Rob, G0VTQ

41. [HFSIG:1429] Re: SS on HF bands

From: Robert.Glassey@nmp.nokia.com
Date: Mon, 5 Aug 96 16:01:39 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022c0ae2@MHS>
Rendered body type: text/plain

> I guess it boils down to what you think ham radio is all
> about. Reliable communications? That description hardly fits direct
> long-haul HF. Easily accessible to those without large lots, liberal
> housing covenants and money for big antennas and power amplifiers? I
> think not. The popularity of VHF/FM and repeater networks shows there
> are a lot of hams who are interested primarily in reliable local
> communications with small stations, and who aren't purists when it
> comes to using relays.

Very true, but then their operation does not lessen the operation or
experimentation of others. Please remember that experimentation for the
purpose of self training is not just for the expert, but also for the
beginner who wants to build a home brew QRP rig, or get a simple RTTY
hack going, or play around with antennas and station configurations to
achieve the best possible propagation under difficult conditions. The
beginner is not up to the level of high efficency systems, he is more
concerned with the effects of SWR on his transmission line, and why his
low vertical performs better than his low dipole, and how much better a
quad might be and how they work. Trying to work 100 countries is a great
incentive to try and understand these things. This is valid
experimentation, and contributes to the understanding of radio
techniques amounst people who are not 'rocket scientists'. By all means
the experts should challange the new frontiers, but not by treading on
the other folks who have less ambitious aims.

Some have critisized traditional narrow band operaters, claiming that
they consider the whole band to be theirs, but really they only
occasionally claim a few kHz, without upsetting anyone else. They see a
few SS zealoites as attempting to hijack the *entire* band as their own,
causing interference and lessening the enjoyment of everyone else.

> where we trail the commercial and military world. While I don't really
> have anything against DXing and contesting, I really don't see how
> they help justify ham radio's existence to the outside world; they're
> really little more than technological stamp-collecting.

see above.

> And whatever risks there are will be worth the benefits. After all,

Who's cost, who's benifit. Not everyone thinks SS is the 'one true
mode', or needs its efficency. Others have other interests.

By all means experiment but not to the detriment of others, its not your
band either. As they say, 'Fools rush in...' and all that. We can start
on a much smaller scale. This is not an ideal world, only the real one.

My weekend experiments have convinced me that I would not like an SS
operator hopping over the entire 20m band, running more than 100
microwatts within 5 miles of my QTH.

Rob.

42. [HFSIG:1436] Re: HF SS Demonstration

From: Robert.Glassey@nmp.nokia.com
Date: Mon, 5 Aug 96 16:46:43 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022c4002@MHS>
Rendered body type: text/plain

> Odd. I find stations tuning up--even weak heterodynes--a lot more
> annoying than SSB splatter or the occasional solar noise burst. The
> San Diego county noise ordinance lowers the one-hour average SPL noise
> limit by 5 dB if the noise contains a steady, audible tone "such as a
> whine, screech, or hum."

Yeah, I know what you mean. I tried a 2ms burst, 10ms burst and a steady
carrier. The stready carrier was easilly the most audible, and the 10ms
burst was worse than the 2ms burst. Compared to the 2ms burst the noise
burst (30ms) was more annoying.

Cheers,

Rob

43. Re: Amateur culture

From: sivula@NCSNRS01ES.ntc.nokia.com (Sivula Timo (NTC/Espoo))
Date: Mon, 05 Aug 1996 13:56:51 +0300
To: hfsig@tapr.org (hfsig)
Message-ID: <1996Aug05.120248.1604.258127@ntcit-mmta6.ntc.nokia.com>
Rendered body type: text/plain

Phil writes:

>Indeed, ham radio doesn't exist in a vacuum. It is not just a hobby.
>It is given spectrum that could otherwise be sold for big bucks
>because it supposedly gives something back to society as a whole. Now
>that emergency communications and international good will have largely
>been taken over by cellular telephones and the Internet, respectively,
>what's left?

Well, what about its educational influence? I do not know whether this
applies
internationally, please advice me, but in Finland the amateur hobby has
very
often lead starting technical studies. In this way I think that society has
got something
back from the giving away of spectrum. I do not think it is a coinsidence
that many
of our colleagues are hams. The hobby is of course just one of the reasons.
Anyway
there is a correlation between spectrum allocated for amateurs and the
amount of
radio engineers, I think.

An other area where society has gained from the hobby is in defence. In
small countries
that cannot afford an expensive professional army, self educated troops are
very valuable. Very often Radio Amateurs only need little extra training

for
becoming good military radio operators and radio officers.

>Technical experimentation and self-training! How much of that
>really goes on in ham radio these days? Not much

I would like to disagree on this point, as well. There is a lot of self
training and experimenting
going on, on the individual level. A normal HF SSB QSO from Finland to the
USA requires a
lot of know how. Everything is of course well documented and existing old
information,
but for the unexperienced individual newcomer HAM this qso is extremenly
self educating
and it is experimental for him/her, as a person. For me, the assembling of
my 7 MHz QRP
XTAL CW rig, a few years ago, was self educating and experimental. As is
now the testing
and writing of code for the DSP4

IMHO we should not restrict the evaluation of the amateur radio hobby to the
absolute edge
of development only. Other areas can be valuable for the society, as well.

ObHFsig: Is there anybody in the US or outside Europe that would be
interested in testing
SP9VRC´s modem software in a QSO to Finland? The software run on a motorola
EVM or
an Alef Null DSP4. I would be interested in testing the slow BPSK modem and
the faster DQPSK
modems in a DX QSO. Required HW is a HF capable amateur station and an EVM
or DSP4
and a PC.

Has anybody any experience of writing a computer control code for the Yaesu
FT 757 GX?
My code for the DSP4 most probably would gain from hearing some experience
from others...

Timo, OH6KK & OH2LVZ

Timo.Sivula@ntc.nokia.com

44. SS on HF

From: Alan Bloom <alanb@polecat.sr.hp.com>
Date: Mon, 5 Aug 1996 11:59:53 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <199608051859.AA259141593@polecat.sr.hp.com>
Rendered body type: text/plain

Phil Karn <karn@unix.ka9q.ampr.org> wrote:

>I agree, and actually that's why I would support dividing up the band
>into "DX" and "regional" subbands to mitigate the near-far problem.
>Each subband would still allow wideband modulation within it, but only
>for the right purpose (i.e., DX or local). I think this would go a
>long way towards solving the problem, don't you?

No. There is no way to separate local from DX stations on HF. Local
stations in the "local" band will be drowned out by other "local"
stations thousands of miles away. And DX stations in the "DX" band
will be munged by other DX stations (some close by and some far away).

>... If everyone's power is matched at the receiver, you can tolerate as
>many as 31.6 other users before their aggregate interference brings
>the Eb/N0 down to 6 dB. ...

To assume that everyone's power is the same is a reasonable first
approximation for a VHF cellular system (give or take 10 dB or so).
But on the HF bands, there are huge differences between signal levels.
It would be rare to be working a station that is within 10 dB or so of
the strongest signal on the band.

>>Spread spectrum works fine for something like a cellular telephone system
>>where the locations and power levels of all stations are well-controlled.
>>But on the amateur bands, stations are randomly distributed.
>
>Most cellular users are randomly distributed too. But they do generally
>talk only to the nearest base station.

You obviously know much more about cellular communications than I do,
but as I understand it:

Since uplink and downlink frequencies are separated by 45 MHz, the mobile
stations will not interfere with each other. Since the mobile stations'
power is controlled by the base stations, mobile stations don't interfere
with base stations. And since mobile stations always talk to the closest
base station, base stations don't interfere with mobile stations.

In all cases, interfering signals are never very much stronger than the
desired signals. A cellular system is a carefully-designed system with
tight controls on location and power levels of transmitters. It also
depends on repeatable propagation characteristics. None of those
conditions exist on the HF amateur bands.

>This is true as far as it goes, but 100 dB adjacent channel rejection
>seems a little generous. 60 dB is a more reasonable estimate, I think.
>Whatever.

Good HF communications receivers these days can have 130 dB or more of
blocking dynamic range for interfering signals 20 kHz away. "Adjacent
channel rejection" (3 kHz away?) would be less than that, but 100 dB
seems a pretty conservative typical number. A receiver with only
60 dB of dynamic range would be horrible.

>>Cellular-style communications doesn't work on DX bands, because
>>nearby stations are typically very much weaker than distant stations.
>>(Because ground wave propagation is much lossier than sky wave.)
>
>Ah, but wouldn't this tend to mitigate the near-far problem?

It's not a "near-far" problem so much as a "strong-weak" problem.
On HF, the strongest stations are rarely the closest ones. At a given
time, a station in Tokyo can easily be 60 dB louder than a similar
station in Denver.

>... That means keeping up with the state of the art in the
>non-amateur world, and spread spectrum is certainly one technology
>where we trail the commercial and military world.

I agree with that. But we need to be sure we don't apply commercial
technology inappropriately. To some extent, that's what happened with
AX.25 packet radio. We took a system that was designed for wire LANs
and adopted it with few changes for radio. It works, but not as well
as it should.

AL N1AL

45. Re: [HFSIG:1438] Re: SS on HF bands

From: Greg Jones <wd5ivd@tapr.org>
Date: Mon, 5 Aug 1996 15:33:50 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608052033.PAA01229@tapr.org>
Rendered body type: text/plain

>
> My weekend experiments have convinced me that I would not like an SS
> operator hopping over the entire 20m band, running more than 100
> microwatts within 5 miles of my QTH.
>

Hello.

I don't think you should make a choice between 'good' or 'bad' based on your
overly simplistic and not very scientific experiment.

Doing these types of analogy testing is good and can help people understand
what is at work here -- but there are certain standards required when doing
these types of testing to give them high trustworthiness and creditability.

Cheers - Greg, WD5IVD

46. Re: [HFSIG:1440] Re: Amateur culture

From: Mike Curtis <wd6ehr@kaiwan009.kaiwan.com>
Date: Mon, 5 Aug 1996 14:15:10 -0700 (PDT)
To: hfsig@tapr.org

Message-ID: <199608052115.OAA24677@kaiwan009.kaiwan.com>
Rendered body type: text/plain

> Well, what about <amateur radios> educational influence?

I can't speak about other countries, but in the USA, you only have to memorize test answers to get a license. Unfortunately, that's precisely what MOST new (and upgrading) amateurs do. Amateur radio here is generally not educational - it has been reduced to nothing more than a test of short term memory.

But in any case, education is not part of our rules here, FCC rules part 97. Advancing the state of the art and other technical aspects are the main reasons given for our existence. "International goodwill" and "Trained radio operators" are mentioned once each, out of the 5 reasons given.

-- mike

47. Re: [HFSIG:1442] Re: SS on HF bands

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 5 Aug 1996 15:39:46 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608052239.PAA29230@servo.qualcomm.com>
Rendered body type: text/plain

Greg,

Actually, I think Robert's experiments are a good start. You may quibble with the details of the experiments, but it's hard to argue that at least trying some actual tests can't add some light to the heat of the discussion. They should be encouraged. And I thought simulating just a narrowband "piece" of the spread spectrum signal was a clever idea.

The one comment I have to Robert is this: yes, you may have been able to detect your signals given that you knew they were there and you were specifically looking for them. But being able to simply detect the simulated SS signal (especially on a quiet band) is not the same as saying they'd interfere with your ability to communicate under typical conditions. Would you have been able to detect them had there been a typical SSB signal on the channel in question, and if you had would it have caused any appreciable interference? That's the real issue here.

Phil

48. Re: [HFSIG:1444] Re: SS on HF bands

From: Greg Jones <wd5ivd@tapr.org>
Date: Mon, 5 Aug 1996 18:18:44 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608052318.SAA08883@tapr.org>
Rendered body type: text/plain

I wasn't trying discourage such work -- that just making a judgement to the mode as being good or bad based off of one small test is not correct.

There is more involved, as you point out below, to the entire process.

I hope we can continue to foster such work and development, but not make judgements based on first impressions.

Cheers - Greg

>
> Greg,
>
> Actually, I think Robert's experiments are a good start. You may
> quibble with the details of the experiments, but it's hard to argue
> that at least trying some actual tests can't add some light to the
> heat of the discussion. They should be encouraged. And I thought
> simulating just a narrowband "piece" of the spread spectrum signal was
> a clever idea.
>
> The one comment I have to Robert is this: yes, you may have been able
> to detect your signals given that you knew they were there and you
> were specifically looking for them. But being able to simply detect

> the simulated SS signal (especially on a quiet band) is not the same
> as saying they'd interfere with your ability to communicate under
> typical conditions. Would you have been able to detect them had there
> been a typical SSB signal on the channel in question, and if you had
> would it have caused any appreciable interference? That's the real
> issue here.
>
> Phil
>
>

49. RE: [HFSIG:1443] Re: Amateur culture

From: Jon Bloom <jbloom@connix.com>
Date: Mon, 5 Aug 1996 19:28:37 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB8304.4FA5E2C0@jbloom.connix.com>
Rendered body type: text/plain

Mike Curtis[SMTP:wd6ehr@kaiwan009.kaiwan.com] wrote:
>
>> Well, what about <amateur radios> educational influence?
>
>
>I can't speak about other countries, but in the USA, you only have to
>memorize test answers to get a license. Unfortunately, that's precisely
>what MOST new (and upgrading) amateurs do. Amateur radio here is generally
>not educational - it has been reduced to nothing more than a test of short
>term memory.

The educational aspects of Amateur Radio need not, and do not, end with the taking of the test. The license requirements exist to set minimum standards that ensure (in theory) that the licensee can safely and properly operate a station. Period. (Whether the current exams meet that goal is a subject best left to a forum other than HFSIG.) I suspect that most amateurs learn far more after getting licensed than they did to *get* licensed. I know I did.

I should also add that Amateur Radio is used as an adjunct to formal educational curricula in many schools. Its role in education is not limited to self-training.

>But in any case, education is not part of our rules here, FCC rules part 97.
>Advancing the state of the art and other technical aspects are the main
>reasons given for our existence. "International goodwill" and "Trained
>radio operators" are mentioned once each, out of the 5 reasons given.

As is contributing to the advancement of the state of the art. But that's neither here nor there. Nowhere in Part 97.1 is priority given to any of the reasons cited. Amateur Radio's role as a voluntary noncommercial communication service is no lesser a part of 97.1 than its role in contributing to the state of the art. And if the 97.1 paragraphs that refer to "advancing skills in both the communications and technical phases of the art" and "expansion of... trained operators, technicians, and electronics experts" don't make education part of our rules, what would?

In addition to which, we would be foolish indeed to put all of our eggs in one basket, so to speak. Fulfilling a number of roles gives a much stronger case than relying on one--or one "main"--role as a rationale for our existence.

As it applies to the current HFSIG discussions, I think that part of our educational process should include investigating new techniques and technologies. On that basis, I'd love to see some exploration of SS techniques on HF. But since nowhere in Part 97 is the rationale for Amateur Radio stated as the development of a maximally efficient communication system, it's not a given--at least to me--that we should abandon all narrowband activity in favor of an SS system even if such a system can be shown to be more spectrum-efficient in theory and practice. Which means that any SS system must coexist with narrowband uses of the spectrum by whatever means make that possible.

-- Jon

50. Probability question

From: Richard Lay <lay@cod.nosc.mil>
Date: Mon, 5 Aug 96 16:30:03 PDT
To: hfsig@tapr.org

Message-ID: <9608052330.AA27592@marlin.nosc.mil>
Rendered body type: text/plain

This question has to do with HF ALE radios and their link establishment protocol. One radio sends out a sequence of ALE words. If the receiver finds these words 'appropriate' or 'correct' from its point of view, it will reply, etc. and the link is established. If the receiver detects so many 'inappropriate' or 'incorrect' or 'bad' ALE words in a row, it will disestablish the link and continue scanning. Now, if the probability that one received ALE word is incorrect is = P, and the sequence of ALE words is S long, then what is the probability that _at least_ Y words _in a row_ are incorrect? It's the "at least" and "in a row" that are giving me trouble!

Rich

51. How much QRM is too much?

From: k6sti@n2.net (Brian Beezley)
Date: Mon, 5 Aug 1996 17:35:57 -0700
To: hfsig@tapr.org
Message-ID: <199608060035.RAA13794@ravel.n2.net>
Rendered body type: text/plain

>The one comment I have to Robert is this: yes, you may have been able
>to detect your signals given that you knew they were there and you
>were specifically looking for them. But being able to simply detect
>the simulated SS signal (especially on a quiet band) is not the same
>as saying they'd interfere with your ability to communicate under
>typical conditions. Would you have been able to detect them had there
>been a typical SSB signal on the channel in question, and if you had
>would it have caused any appreciable interference? That's the real
>issue here.

For someone trying to copy a signal right at the thermal, cosmic, or atmospheric noise level, any interference is appreciable. For some hams (particularly 160m, 6m, and VHF/UHF weak-signal types but also for some HF DXers and QRPers), trying to make two-way contact with extremely weak signals constitutes a large part of their amateur activities. These people will never welcome power that could have been put elsewhere, even if it raises their noise floor only by a dB or two. An EMEer can turn blue at the thought of losing of a few tenths of a dB!

Brian Beezley, K6STI
k6sti@n2.net

52. Re: [HFSIG:1447] Probability question

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Mon, 5 Aug 1996 21:21:39 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608060221.VAA11427@www.kelly-afb.org>
Rendered body type: text/plain

Rich,

If you really want the Mil-Std protocol I can get it but I think you are basically correct. There are a couple things that makes ALE not really desirable for hambands...one is that the query transmitter transmitts a _long_ (very long) series of "words" at a very low baud rate...its called "order wire". The length is long enough for the desitnation/desired receiver to scan _all_ the channels in the ALE program. If the "network" has 8-12 channles the scan time for the Mil-Std ALC could be over a minute...this means that the quesy transmitter would transmit for over a minute. Talk about QRM...and to make matters worse, the Mil-Std ALE protocol doesn't listen on a channel before it transmitts...if you have an established communication (QSO) going on, it will walk right on top of you. We disabled it in Saudi during the Gulf War ('90-'91) in favor of doing a daily propagation study using Minimuf...this worked better.

The Mil-Std and Fed-Std have been pushed by several hams thru the ARRL and there was even an big article in QST a couple of years ago.

73, Walt

In your message you write:
>
> This question has to do with HF ALE radios and their link establishment

> protocol. One radio sends out a sequence of ALE words. If the receiver
> finds these words 'appropriate' or 'correct' from its point of view, it will
> reply, etc. and the link is established. If the receiver detects so many
> 'inappropriate' or 'incorrect' or 'bad' ALE words in a row, it will
> disestablish the link and continue scanning. Now, if the probability that
> one received ALE word is incorrect is = P, and the sequence of ALE words is
> S long, then what is the probability that _at least_ Y words _in a row_ are
> incorrect? It's the "at least" and "in a row" that are giving me trouble!
>
> Rich
>
>

--
```
========================================================================
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development.      |
========================================================================
|                                  |                                   |
|                                  | The greatest dangers to liberty   |
| Walt DuBose - K5YFW              | lurk in insidious encroachment    |
| E-Mail k5yfw@www.kelly-afb.org   | by men of zeal, well-meaning      |
| Business Telephone: (210)925-6081 | but without understanding.       |
|     Home Telephone: (210)696-3196 |                                   |
|                                  |   - Justice Louis D. Brandeis     |
|                                  |                                   |
========================================================================
```

53. Re: [HFSIG:1446] Re: Amateur culture

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Mon, 5 Aug 1996 21:42:23 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608060242.VAA11463@www.kelly-afb.org>
Rendered body type: text/plain

Jon, et al,

I've remained silent too long on this subject.

I have to agree that most hams do most certainly learn more after they
receive their "first" license. Few folks in the U.S. go directly from
being un-licensed to Extra Class. When a ham stops learning, they should
turn in their license.

There is a ham, and perhaps on this SIG, but known to the ARRL Educational
folks (Jon ask Rosalie) who got a big bucks job a with a cellphone company
because of his hands-on "radio" experience through hamradio...even tho
his had not finished his degree...others applying for the job had
degrees, post-grad. degrees, but no hands-on experience. Our hambands
are becomming the only available frequencies that educational
institutions can readly use for experimentation...so all you teachers,
Middle school thru post grad. get a ham ticket.

I am on the Board of Directors of the Young Astronaut Technology Program
in the Northeast School district here in San Antonio and we use ham radio
as a teaching tool.

One other point. Many of us here in San Antonio have had the oppertunity
to "take back" event communications from trunking radios and/or cellphones.
The individual with the radio 1) may not be trained and does a bad job of
communiting 2) doesn't listen to the radio 3) Is too busy taking care of
event business to mess with the radio. Our radio operators are the VIPs
ears and voice. Ah ha, now I remember why in WWII and the Korean
Conflict we had radio operators...and radio operators on ships and
aeroplanes...the NCOIC Officer, Captain, pilot is too busy making
decisions to listen to the radio. Also, cellphones don't talk to a whole
bunch of persons all at once.

73, Walt

 In your message you write:
>
> Mike Curtis[SMTP:wd6ehr@kaiwan009.kaiwan.com] wrote:
> >
> >> Well, what about <amateur radios> educational influence?
> >
> >
> >I can't speak about other countries, but in the USA, you only have to

> >memorize test answers to get a license. Unfortunately, that's precisely
> >what MOST new (and upgrading) amateurs do. Amateur radio here is generally
> >not educational - it has been reduced to nothing more than a test of short
> >term memory.
>
> The educational aspects of Amateur Radio need not, and do not, end with
> the taking of the test. The license requirements exist to set minimum
> standards that ensure (in theory) that the licensee can safely and
> properly operate a station. Period. (Whether the current exams meet that
> goal is a subject best left to a forum other than HFSIG.) I suspect
> that most amateurs learn far more after getting licensed than they
> did to *get* licensed. I know I did.
>
> I should also add that Amateur Radio is used as an adjunct to formal
> educational curricula in many schools. Its role in education is not
> limited to self-training.
>
> >But in any case, education is not part of our rules here, FCC rules part 97.
> >Advancing the state of the art and other technical aspects are the main
> >reasons given for our existence. "International goodwill" and "Trained
> >radio operators" are mentioned once each, out of the 5 reasons given.
>
> As is contributing to the advancement of the state of the art. But that's
> neither here nor there. Nowhere in Part 97.1 is priority given to any of
> the reasons cited. Amateur Radio's role as a voluntary noncommercial
> communication service is no lesser a part of 97.1 than its role in
> contributing to the state of the art. And if the 97.1 paragraphs that
> refer to "advancing skills in both the communications and technical
> phases of the art" and "expansion of... trained operators, technicians,
> and electronics experts" don't make education part of our rules, what
> would?
>
> In addition to which, we would be foolish indeed to put all of our eggs
> in one basket, so to speak. Fulfilling a number of roles gives a much
> stronger case than relying on one--or one "main"--role as a rationale
> for our existence.
>
> As it applies to the current HFSIG discussions, I think that part of
> our educational process should include investigating new techniques and
> technologies. On that basis, I'd love to see some exploration of SS
> techniques on HF. But since nowhere in Part 97 is the rationale for
> Amateur Radio stated as the development of a maximally efficient
> communication system, it's not a given--at least to me--that we should
> abandon all narrowband activity in favor of an SS system even if such
> a system can be shown to be more spectrum-efficient in theory and
> practice. Which means that any SS system must coexist with narrowband
> uses of the spectrum by whatever means make that possible.
>
> -- Jon
>
>

--
==
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development. |
==
	The greatest dangers to liberty
Walt DuBose - K5YFW	lurk in insidious encroachment
E-Mail k5yfw@www.kelly-afb.org	by men of zeal, well-meaning
Business Telephone: (210)925-6081	but without understanding.
Home Telephone: (210)696-3196	
	- Justice Louis D. Brandeis
==

54. Re: [HFSIG:1442] Re: SS on HF bands

From: Charles Hoffman <choffman@pelican.davlin.net>

Date: Mon, 05 Aug 1996 21:57:22 -0500
To: hfsig@tapr.org
Message-ID: <3206B492.2E70@pelican.davlin.net>
Rendered body type: text/plain

Gentlemen, please! May I make an humble and polite suggestion? After
some 37 years of observations of amateur radio, military and commercial
technological pioneering, and as a former university professor
responsible for raising the awareness of my students-in-ernest, like

yourselves, please permit me to suggest that it is always important to
do your research. See what has already been done. Experiments with
interference is history, check your technical library, do your homework
so to speak and beyond all, do not invent with a thesauraus in your left
hand. All good ideas, but everything I have read here for some time has
already been done. Ask the questions and get the answers, then go from
there. SS is over twenty years here, eh? Perhaps you might check with
W3XO and some of the much earlier pioneers in SS to see what they have
become bored with. HO HO!

Everyones efforts are recognized and appreciated greatly, believe me,
and I am speaking for myself, but I am reading great ideas (seriously)
but which have already been invented. Read your resources and save your
greatest creativity for that which has not yet been tried, concluded;
focus, based upon not what has already been done, but amplify from there
with your refreshing creations of cogitation.

This is meant to be complimentary, but I am becoming somewhat frustrated
with the thread (perhaps, "been there done that", will mean something in
the popular venacular--no offense intended 8-o). Please carry on
gentlemen, you have my undivided attention. We are all one anothers
teachers; I do hope my decades of experience is somehow beneficial as we
are indeed all on the same tangent here. WEll done to all.

73 Rik de K5SBU

Greg Jones wrote:
>
> >
> > My weekend experiments have convinced me that I would not like an SS
> > operator hopping over the entire 20m band, running more than 100
> > microwatts within 5 miles of my QTH.
> >
>
> Hello.
>
> I don't think you should make a choice between 'good' or 'bad' based on your
> overly simplistic and not very scientific experiment.
>
> Doing these types of analogy testing is good and can help people understand
> what is at work here -- but there are certain standards required when doing
> these types of testing to give them high trustworthiness and creditability.
>
> Cheers - Greg, WD5IVD

55. Re: [HFSIG:1449] Re: Probability question

From: Charles Hoffman <choffman@pelican.davlin.net>
Date: Mon, 05 Aug 1996 22:57:40 -0500
To: hfsig@tapr.org
Message-ID: <3206C2B4.64C@pelican.davlin.net>
Rendered body type: text/plain

100% on this one guys...Walt is on target here; forget about ALE on
amrad. First we are not appropriately authorized. Second, the
propogation avenues are not appropriate for ALE on the linear harmonic
related freqs we have available. This system is designed for total HF
access. Lets refocus here.

Walt DuBose - K5YFW wrote:
>
> Rich,
>
> If you really want the Mil-Std protocol I can get it but I think you are
> basically correct. There are a couple things that makes ALE not really
> desirable for hambands...one is that the query transmitter transmitts a
> _long_ (very long) series of "words" at a very low baud rate...its called
> "order wire". The length is long enough for the desitnation/desired
> receiver to scan _all_ the channels in the ALE program. If the "network"
> has 8-12 channles the scan time for the Mil-Std ALC could be over a
> minute...this means that the quesy transmitter would transmit for over a
> minute. Talk about QRM...and to make matters worse, the Mil-Std ALE
> protocol doesn't listen on a channel before it transmitts...if you have an
> established communication (QSO) going on, it will walk right on top of
> you. We disabled it in Saudi during the Gulf War ('90-'91) in favor of
> doing a daily propagation study using Minimuf...this worked better.
>
> The Mil-Std and Fed-Std have been pushed by several hams thru the ARRL
> and there was even an big article in QST a couple of years ago.
>

> 73, Walt
>
> In your message you write:
> >
> > This question has to do with HF ALE radios and their link establishment
> > protocol. One radio sends out a sequence of ALE words. If the receiver
> > finds these words 'appropriate' or 'correct' from its point of view, it will
> > reply, etc. and the link is established. If the receiver detects so many
> > 'inappropriate' or 'incorrect' or 'bad' ALE words in a row, it will
> > disestablish the link and continue scanning. Now, if the probability that
> > one received ALE word is incorrect is = P, and the sequence of ALE words is
> > S long, then what is the probability that _at least_ Y words _in a row_ are
> > incorrect? It's the "at least" and "in a row" that are giving me trouble!
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> > Rich
> >
> >
>
> --
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> | The MicroSoft operating system didn't get as bad as it is overnight,|
> | it has taken over 10 years of careful, calculated development. |
> ==
> | | |
> | | The greatest dangers to liberty |
> | Walt DuBose - K5YFW | lurk in insidious encroachment |
> | E-Mail k5yfw@www.kelly-afb.org | by men of zeal, well-meaning |
> | Business Telephone: (210)925-6081 | but without understanding. |
> | Home Telephone: (210)696-3196 | |
> | | - Justice Louis D. Brandeis |
> | | |
> ==

56. Re: [HFSIG:1446] Re: Amateur culture

From: Mike Curtis <wd6ehr@kaiwan009.kaiwan.com>
Date: Tue, 6 Aug 1996 00:14:34 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608060714.AAA17899@kaiwan009.kaiwan.com>
Rendered body type: text/plain

>
> Mike Curtis[SMTP:wd6ehr@kaiwan009.kaiwan.com] wrote:
> >
> >> Well, what about <amateur radios> educational influence?
> >
> >
> >I can't speak about other countries, but in the USA, you only have to
> >memorize test answers to get a license. Unfortunately, that's precisely
> >what MOST new (and upgrading) amateurs do. Amateur radio here is generally
> >not educational - it has been reduced to nothing more than a test of short
> >term memory.
>
> The educational aspects of Amateur Radio need not, and do not, end with
> the taking of the test. The license requirements exist to set minimum
> standards that ensure (in theory) that the licensee can safely and
> properly operate a station. Period. (Whether the current exams meet that
> goal is a subject best left to a forum other than HFSIG.) I suspect
> that most amateurs learn far more after getting licensed than they
> did to *get* licensed. I know I did.
>
> I should also add that Amateur Radio is used as an adjunct to formal
> educational curricula in many schools. Its role in education is not
> limited to self-training.

I agree that a lot of hams learn more after being licensed than to be
licensed. But my points were that 1. to GET the license requires ONLY rote
memorization, and 2. unfortunately MOST hams never progress much beyond what
is required.

Both of us have taken the lead in education among hams. And there have been
(and will be) a lot of hams who are interested in learning what they really
should know for the license - and beyond. Unfortunately, many of the
ignoramuses who are just interested in appliance operation make things
difficult. There used to be a popular repeater here in LA where technical
talk was "officially" discouraged. I don't know if it's still "no-tech", or
even if it's still on the air, but that's unimportant.

> As it applies to the current HFSIG discussions, I think that part of
> our educational process should include investigating new techniques and

> technologies. On that basis, I'd love to see some exploration of SS
> techniques on HF. But since nowhere in Part 97 is the rationale for
> Amateur Radio stated as the development of a maximally efficient
> communication system, it's not a given--at least to me--that we should
> abandon all narrowband activity in favor of an SS system even if such
> a system can be shown to be more spectrum-efficient in theory and
> practice. Which means that any SS system must coexist with narrowband
> uses of the spectrum by whatever means make that possible.

One of the best things about amateur radio IMHO is its diversity. I think most of us will agree with this wholeheartedly. Nowhere have I seen a proposal to pursue any particular mode to the exclusion of all others. So I don't see abandonment of all narrowband technology as anything to be concerned about.

Over the years, we've had to make adjustments in band usage. When RTTY became more popular, we made room for it. When SSTV came around, we adjusted. Ditto packet, AMTOR, et al. But all the "old" modes are still around (well, except spark, but that's not the fault of anything other than the mode itself :-)

One thing for sure - it's going to be most interesting watching things unfold.

-- mike

57. EVM56002 ?

From: Greg Jones <wd5ivd@tapr.org>
Date: Tue, 6 Aug 1996 04:12:31 -0500 (CDT)
To: hfsig@tapr.org (HF SIG mailing)
Message-ID: <199608060912.EAA05283@tapr.org>
Rendered body type: text/plain

There is an opportunity for TAPR to do a group purchase on the Motorola EVM56002 at a pretty low cost. Between 20 and 200 units would be the goal...same price no matter how many are purchased. Figure about $60 under current commercial prices. No hard prices, becuase more work would have to be done.

Would there be interest in this in the HF SIG ? Espiecally since Johan's code would run on it.

I have spoken with Johan and he thinks it would be great...what are some other reactions.

Cheers - Greg, WD5IVD

58. [HFSIG:1442] Re: SS on HF bands

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 6 Aug 96 10:16:32 +0100
To: hfsig@tapr.org, wd5ivd@tapr.org
Message-ID: <H0000292022d7982@MHS>
Rendered body type: text/plain

> >
> > My weekend experiments have convinced me that I would not like an SS
> > operator hopping over the entire 20m band, running more than 100
> > microwatts within 5 miles of my QTH.
> >
>
> Hello.

> I don't think you should make a choice between 'good' or 'bad' based
> on your overly simplistic and not very scientific experiment.

Err, wait a moment, I'll post my report today hopefully, it certianly isn't the be all to end all of propagation studied, but a considerable amount of effort was put into this, and extra special attention was paid to minimising the level of interference a SS signal could generate, including trying various different pulse shapes and power levels.

> Doing these types of analogy testing is good and can help people
> understand what is at work here -- but there are certain standards
> required when doing these types of testing to give them high
> trustworthiness and creditability.

I hope that my experiments have been done to a standard that will give

the results a respectible degree of trustworthyness and credibility.
Considerable effort was put into calibration and characterisation of
both TX and RX. There is of course a large error margin (either way) on
the results (mostly due to the terain), however this was qualified and
considered in context of the severity of the interference observed and
enviromental factors.

The report will be out soon.

Cheers,

Rob

59. [HFSIG:1444] Re: SS on HF bands

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 6 Aug 96 10:37:50 +0100
To: hfsig@tapr.org, karn@qualcomm.com
Message-ID: <H0000292022d798d@MHS>
Rendered body type: text/plain

> The one comment I have to Robert is this: yes, you may have been able
> to detect your signals given that you knew they were there and you
> were specifically looking for them. But being able to simply detect
> the simulated SS signal (especially on a quiet band) is not the same
> as saying they'd interfere with your ability to communicate under
> typical conditions.

Yes, listening for the signal does make it easier to hear, but I did not
consider 'maybe' interference significant, only when clear tone or noise
bursts could be heard. While you may not notice this when briefly
listening to a signal, any such signals will become much more apparent
when you listen for a longer time, especially if you are listening to a
weak station. Typical conditipons are not relevant, since this is not
the area of concern, rather the concern is over the interference caused
to signals that are already dificult to hear, caused by a hopping
transmitter within a few miles.

> Would you have been able to detect them had there
> been a typical SSB signal on the channel in question, and if you had
> would it have caused any appreciable interference? That's the real
> issue here.

As it happens, 18MHz was quite an active band during my tests, and there
were times when even on my 70cm long whip antenna a strong SSB signal
came through strongly. In my tests I compared a constant carrier level
with amateur and broardcast station levels, and pulsed interference with
a constant carrier. My scanner with the whip antenna was not capable of
receiving weak signals, being approximatly 30dB less sensitive that my
dipole at home, in the presence of external noise. Still, the
interference could be heard although very strong amateur stations could
overpower it.

More details in the report...

Cheers,

Rob

60. Re: [HFSIG:1454] EVM56002 ?

From: Mike Murphree <n4cnw@pig.net>
Date: Tue, 06 Aug 1996 07:47:08 -0400
To: hfsig@tapr.org
Message-ID: <320730BC.1500@pig.net>
Rendered body type: text/plain

Greg Jones wrote:

> There is an opportunity for TAPR to do a group purchase on the Motorola
> EVM56002 at a pretty low cost. Between 20 and 200 units would be the
> goal...same price no matter how many are purchased. Figure about $60 under
> current commercial prices. No hard prices, becuase more work would have to
> be done.
>
> Would there be interest in this in the HF SIG ? Espiecally since Johan's
> code would run on it.

Sounds great considering the price is already only $149.95 at the
Design-Net price page (http://www2.motorola-dsp.com/dsp/home/net/pg.html).

They also offer a student discount of 20% reducing the price to $120 + $3
shipping (http://www2.motorola-dsp.com/dsp/home/edu/discount.html)

Mike N4CNW

61. Amateur culture and not so wide modes

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 6 Aug 96 16:04:00 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022d7998@MHS>
Rendered body type: text/plain

Hi Mike, good to hear another voice.

> I agree that a lot of hams learn more after being licensed than to be
> licensed. But my points were that 1. to GET the license requires ONLY
> rote memorization, and 2. unfortunately MOST hams never progress much
> beyond what is required.

True, but I have no problem with this. Nobodys lost out even if they
never pick up a mic. At least they have explored an interest, even
learnt something about radio, however little. Even appliance operators
can get the itch to try something more, to get a bigger antenna, to try
and work the repeater 50 miles away to talk to their friends over there,
for many that small interest can develope into much more. One of the
great things about amateur radio is that there is always more to try, no
matter how far you get involved there is always more chalanges,
something new and interesting. Amateur radio encourages an understanding
of radio in the wider community, to lift some of the mystery of this
'black art'. It might even help people understand how to tune their
video receoders if their friend or relative is a ham :-) But seriously,
ham radio is for everyone, expert, beginner, emergency operators, and
even the appliance operator.

> One of the best things about amateur radio IMHO is its diversity. I
> think most of us will agree with this wholeheartedly. Nowhere have I
> seen a proposal to pursue any particular mode to the exclusion of all
> others. So I don't see abandonment of all narrowband technology as
> anything to be concerned about.

I hope not, just to reprint the few lines that started this thread off:

On 24th July Phil wrote:

> But it's precisely those 300 baud/1 KHz restrictions that I want to
> eliminate. Eventually I want to spread over the whole band with
> frequency hopping, but to start I'd like to do MFSK over SSB
> bandwidths.

This was said in the context of 'how much bandwidth can we hop over' and
the opinions and attitudes that Costas called 'Narrow Band Thinking' in
his classic paper (which I'm sure Phil has framed on his wall :-).

The idea here was simply to use an SSB channel. The previous discussion
involved whether narrow or wide modes were better.

I personally go along with Phil to an extent. I like the idea of trying
this mode out and developing the technology it uses. But while there may
be gains in spreading over the whole band, those gains are for only a
few, and may well be to the detriment of all other users. I am in favour
of say a 10-25KHz allocation for all wide band channel sharing modes. I
see considereable scope here for experimentation, and to overcome the
problems of HF packet. (CSMA sharing using 300 baud on HF would be
hillarious if people didn't actually try to use it!) It needs to be
replaced, and this may be the answer. We need to get clever on this one,
and not just rely on spreading to overcome interference to the users of
this mode by lumping it on everyone else. I think the answer lies in
coding and inteligent use of redundancy, combined with a small degree of
hopping to use the trasciever filters to counter the few very strong
signals (make them random 'hits'). 25kHz would be heaps, and if you dont
need to worry about sharing with other modes, then you have much more
freedom to try different techniques, without the constraints of
non-interference to narrow band DX stations.

This could be thought of as not so much 'spread spectrum' but more a
wide Code Division Multiple Access (CDMA) mode to replace the current
wide (total spectrum usage) carrier sense (CSMA) HF packet system.
(shades of "it's not a bomb" for those few New Zealand readers :-)

Sorry Phil, I think maybe I've overreacted a bit here, especially since your original comment was more to do with lifting the 1kHz 300 Baud restrictions in the US, and using an SSB channel, which I would support anyhow. Its spreading over the whole band that I and many others oppose. Maybe we've come full circle here, but I think a few good comments have come out.

Cheers,

Rob

62. Re: [HFSIG:1457] Re: EVM56002 ?

From: forrerj@peak.org (Johan Forrer.)
Date: Tue, 06 Aug 1996 08:31:19 -0800
To: hfsig@tapr.org
Message-ID: <199608061542.IAA23152@PEAK.ORG>
Rendered body type: text/plain

Hi All,

Let me just recap what the possibilities are:

- A 60 MHz 56002, concervatively rated. Some have shown to run reliably at 80 MHz. (When executing multiple instructions per clock tick, this effectively translates
  to 60*3=180 MIPS or at a minimum, one instruction per clock tick - 60 MIPS).
  This is a 24-bit machine (120 dB dynamic range); for comparison, its TWICE as fast as the 16-bit DSP chip as used in the new Pactor II modem.

- The 56K instruction set looks a lot like Motorola's other microcontrollers, however, is a dual Harward machine and a true DSP. It's a pleasure to write code
  for it.

- Has an on-board serial peripheral, some spare I/O lines and adequate memory for
  doing innovative programming projects.

- Comes with a very decent, full-screen, multiple windows, debugger that was developed
  by Domain Technologies. The debugger software is stable and has undergone several
  revisions giving you the benefit of user-feedback. The debugger is OnCe-based, i.e.,
  uses special on-chip silicon to do non-intrusive debugging at full DSP speeds.

- A decent assembler with macro facilities. Public domain C compiler, linker, librarian and simulator is available.

- A port of the Alef Null monitor "Leonid" bridges the EVM and the Finnish DSP Card 4.
  This gives you access to some outstanding software efforts.

- A substantial amount of amateur radio-related software including modems for packet,
  WEFAX, satellite work, and of course HF digital. To get on the air, you need the EVM and a PC that talks the KISS protocol. Several options are possible.

I own two EVM's myself that have been adapted for amateur radio use. One is configured as a single port radio modem and the other as a dual-port modem - it has expanded memory to 32K words in each of the P, X, and Y memory spaces. When I find the time, I hope to finish an article on the interface for this little beast. One of my future projects is to port the popular multitasking OS called uC O/S (micro-C O/S) to this expanded EVM. It would greatly simplify the development of multithreaded code, especially the kind of thing that is needed when implementing communications protocols. The EVM has plenty of spare horsepower to handle such a task in addition to real-time modems.

Anyone is interested in learning about DSP and am ready to tinker with it at the hardware level, this is a great opportunity. Be assured that the parts on the EVM alone is worth more than TAPR's asking price.

Forgive me if I sound like a Motorola sales person - I don't work for them, just am excited about this bit of hardware and want to make sure that you don't miss up on a good opportunity.

--Johan, KC7WW

>Greg Jones wrote:
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>
>Mike N4CNW
>
>

63. Re: [HFSIG:1447] Probability question

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)
Date: Tue, 6 Aug 96 13:03:31 CDT
To: hfsig@tapr.org
Message-ID: <9608061803.AA01687@eagle.aud.alcatel.com>
Rendered body type: text/plain

> This question has to do with HF ALE radios and their link establishment
> protocol. One radio sends out a sequence of ALE words. If the receiver
> finds these words 'appropriate' or 'correct' from its point of view, it will
> reply, etc. and the link is established. If the receiver detects so many
> 'inappropriate' or 'incorrect' or 'bad' ALE words in a row, it will
> disestablish the link and continue scanning. Now, if the probability that
> one received ALE word is incorrect is = P, and the sequence of ALE words is
> S long, then what is the probability that _at least_ Y words _in a row_ are
> incorrect? It's the "at least" and "in a row" that are giving me trouble!
>
> Rich
>
>

Rich: I did this rather quickly, but here's a guess:

the probability of Y& words in a row are incorrect should be:

$$(1 - P^Y)^(S+1-Y) \qquad (1)$$

the probability of Y or more words in a row are incorrect should be:

$$\sum_{k=0}^{(S-Y)} (1-P^[Y+k])^(S+1-Y+k) \qquad (2)$$

Again, I have not spent the time to verify this. If P is
small, then (1) may be sufficiently accurate.

- Tom

Tom McDermott
mcdermot@aud.alcatel.com

64. Pulses carrier propagation tests

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 6 Aug 96 19:26:18 +0100
To: hfsig@tapr.org
Message-ID: <H0000292022ed98b@MHS>

Short Distance Ground Wave Propagation of Low Power Pulsed HF Transmissions

Introduction

A series of tests were conducted to determine the propagation characteristics of low power pulsed transmissions on 18 MHz over a distance of 1 to 5 miles. The aim of this investigation was to determine the level of interference a frequency hopping spread spectrum transmitter would cause to nearby stations.

The Transmitter:

A Yaesu FT890AT transceiver was used, with the RF power control set to minimum. This gave an output power of about 3 Watts in CW mode, and the same in LSB mode when the audio drive was sufficient to cause the ALC to operate.
Power was further controlled by adjusting the audio input level in LSB mode. In this way output power was varied between 1 Watt and 10 mW pep. LSB was used since carrier suppression was best in this mode. The carrier was undetectable at 1/2 a mile using the receiver below.

The Built in ATU was used to tune the TX antenna. The TX antenna was an 80m dipole, 4m above the ground. Other amateur station may well use better antennas for transmission.

The audio source was an 8 bit PC sound card.

The Receiver:

Yupiteru MVT7100 handheld scanner, with a 1m whip antenna, or a simple dipole and balun resonating around 18MHz.

Using the unmatched 1m whip antenna received signal levels were 10dB below the dipole with a 20dB attenuator, in the same position. When the dipole was used the MVT7100's built-in 20dB attenuator had to be used due to the poor intermod performance of the receiver. This the whip was 30dB less sensitive than the dipole.

To account for the external noise floor the SNR of weak signals using the MVT7100 + dipole + 20dB attenuator was compared with the FT890 + same dipole. The SNR from the MVT7100 was about 10dB worse. i.e. the 20dB attenuator results in a 10dB drop in SNR since the noise floor from the dipole is about 10dB above the MVT7100 noise floor without the attenuator. Thus the noise floor drops only 10dB while the signal drops 20dB. The SNR of a typical DX setup with directional antennas (lower noise) high off the ground (stronger signal) is probably about 10 dB or better than the low RX dipole I used. Thus the MVT710 + low dipole + 20dB attenuator is about 10+10 = 20 dB worst than a typical DX setup.

The whip antenna was found to be 10dB worst than the dipole + 20dB attenuator. Since SNR with the attenuator and dipole was not limited by the external noise floor, the whip antenna has 30dB lower SNR than a typical DX setup.

The Terrain:

The surrounding area is hilly, part pine forest, and part a lightly wooded common with some rough open areas. Total wooded area is around 20 square miles. The hills are formed by a the erosion of a plateau a few remaining parts of the flat plateau at the hill tops. The plateau is about 120-130m ASL. The site is 30 miles inland. The TX site is on the edge of the wooded land in a built up area, 121m ASL.

The Tests:

CW tests were done with 3 Watts, on 18.090 MHz. Only he whip was used to determine the extent and characteristics of propagation in this area. The signal was compared with the strength of amateur stations and broadcast stations.

Pseudo Frequency Hopping Spread Spectrum tests were done by transmitting a series of pulsed a carriers at random frequencies and times to simulate a Frequency Hopping Spread Spectrum signal as it would appear in a 2.4 kHz bandwidth. The pulses were generated using shaped audio tone bursts to modulate the transmitter in LSB mode. The LSB suppressed carrier frequency was 18.150 MHz.

A series of different PFHSS tests were done trying different PEP power levels and pulse shapes. A constant carrier was also transmitter for reference at each power level.

Four pulse shapes were used, these are described below.

The timing of the pulses followed a Poisson distribution. The mean pulse rate varied depending upon the test.

Power levels were: 1 Watt, 100mW, 10mW, and a CW ID for each test at 3 Watts.

Pulse 1: Square 2ms pulsed tones, random phase and audio frequency 300-2700Hz. Mean pulse rate was 3.4 pulses per second. (equivilant to frequency hopping spread spectrum as appears in a 2.4kHz bandwidth when spread over 350kHz with a 2ms dwell time.)

Pulse 2: Raised cosine pulsed tone, 4ms total, -6db pulse duration 2ms. Random phase and frequency with the same pulse positioning as 1.

Pulse 3: 10ms pulsed tones, raised cosine edge shaping, 2ms each side, 10ms total pulse length. FHSS equivilant hopping rate would be 1.4 seconds per pulse. The pulse rate used was twice this, so enough pulses could be heard to determine the signal level. Audio frequency and phase were random.

Pulse 4: 30ms modulated pulsed tones. No shaping, but random BPSK modulation at 2000 bps. Equivilant FHSS pulse rate would be 4.4 seconds per pulse. The rate used was 1.1 seconds per pulse. Centre frequency and phase were random.

Each test lasted 20 seconds. The computer was set up to sequence through the four pulse tests plus a CW test, at each power level. Each test and power level had a unique morse ID.

The Results:

CW measurements.

The 3 watt CW signal was easily heard in all locations with the MVT7100 and whip antenna beyond 5 miles. At 5 miles the signal was just audible with the whip antenna and the 20 dB attenuator on.

Typical signal levels at 1 to 1.5 miles on in good DX locations (on hill tops at the same level as the TX) were 10-30dB above the strongest amateur stations on 18MHz and 14MHz (using the attenuator and calibrated S meter), and 10-30dB below the strongest broadcast stations (BBC, Deutchewelle, VOA, VORussia, Radio Canada) The signal strength was effected more by terrain than distance. The strongest broadcast stations were typically 30-40dB above the strongest amateur stations. Tests were done between 8pm and 10pm local daylight saving time (GMT+1) on Friday 2/8/96.

Pulsed tests.

SOUND: Pulses 1,2,3 all had discernible tones. Pulse 1 was least discernible, pulse 2 was slightly more pronounced, pulse 3 had a clearly discernible tone. The modulated tone, pulse 4, sounded like a burst of noise, with a discernible beginning, end and pitch. Pulse streams 1 and 2 sounded like many taps dripping water into many buckets, all with different pitches. Pulse stream 3 were clear tone bursts. On air any of these pulses were easily distinguishable from a static click and would be annoying however infrequent.

AUDIBILITY: Pulses 1 and 2 seemed to be about the same loudness in the presence of noise, and were the softest of the four pulses. Pulse 3 was more audible, and pulse 4 was the clearest. In terms of signal strength, pulses 1 and 2
required 20dB more power than a CW signal to be audible. Pulses 3 and 4 required 10dB more power. (these results came from comparing received

signals with different TX powers).

RANGE:

RX measurements were made between 1 and 3 miles from the TX, with both the whip and dipole antenna. Again signal strength depended more upon terrain than distance, although there was a clear difference between 1 and 3 miles. Distances were measure using an Ordinance Survey Map 1:50,000. (standard British maps) These are pretty good maps.

At good receiving sites (on flat ground or hill tops on the same level as the TX) all tests were audible with 1 Watt TX power and the whip antenna at a range of 1-3 miles.

At some sites the 100mW pulse 1 and 2 tests could not be heard with the whip antenna, but tests 3 and 4 could be heard. Using the dipole antenna and the 20dB attenuator often made the pulse 1 and 2 tests audible in these cases.

At 3 miles only the 10mW carrier could be heard, no other 10mW tests could be heard with the whip antenna.

At 1 mile the 100mW pulse 1 and 2 tests could often be heard with the whip antenna. In one case at a distance of 1 mile, the 10mW pulse 1 and 2 tests were clearly audible with the whip antenna.

Since whip antenna was estimated to be more than 30dB less sensitive than a good amateur station when the external noise floor is considered (see above), thus if 1 watt TX power is heard using the whip antenna then 1mW will be heard by a good amateur station. In the case where 10mW pulse 1 and 2 TX power was heard clearly at 1 mile, a good amateur station would hear a 10uW 2ms carrier burst.

Conclusion:

Signal strengths of a 3 Watt CW transmitter at a distance of 1 mile were considerable greater (10-30dB) than any other amateur stations heard on the band.

Shorter pulse lengths are less audible, despite being more frequent.

Attempting to spread a long (30ms) tone to make it less noticeable did not work.

A 1mW 2ms pulse at 18MHz would be clearly audible above the ambient noise floor by a good amateur station up to 3 miles from the transmitter site.

A 10uW 2ms pulse at 18MHz may well be heard by a good amateur station at a distance of 1 mile.

Interference may be more severe if the TX station has a better antenna than a 80m dipole 4m above the ground.

Possible Further Work:

Interference decreased as the pulse length got shorter, and subjectively the sound was less was annoying. If pulse length was made even shorter, say 0.1mS, then the subjective effect of the pulse may be acceptable. Such a short pulse length would spread power wider than a single SSB channel, further reducing the level of the received pulse, and also making it more compatible with common impulse noise blankers. If such a short dwell time were used in a hopping system, spreading over 350kHz, then the average pulse rate in a narrow receiver bandwidth (<3kHz) would be:

$$\text{Rate} = \frac{\text{pulse\_band\_width}}{\text{total\_band}} / \text{dwell\_time}$$

$$= 20 / 350 / 100\text{E-}6$$

$$= 571 \text{ pulses per second on average.}$$

This would probably be unacceptable, since the average energy in the receiver bandwidth would be high and not easily dealt with. This would probably be very similar to a direct sequence system.

If spread over 15MHz, rate = 13 pulses per second on average.

The ideal dwell time (pulse length) would probably be in the range 0.1
to 0.5 ms. Further investigation would be required.

Further reduction in pulse rate could be achieved by trading off more
power for fewer pulses, if the pulses are short enough to sound
sufficiently noise like.

Interference from such a system may be acceptable.

65. Re: [HFSIG:1459] Re: EVM56002 ?

From: zs6awk@global.co.za (Danie Brynard)
Date: Tue, 06 Aug 1996 22:38:46 +0200
To: hfsig@tapr.org
Message-ID: <199608070036.WAA21477@lin01.global.co.za>
Rendered body type: text/plain

I can only agree: the DSP56002EVM is a wonderfull piece of hardware. I have
been in love with mine for 1.5 years :-)

Newcomers please see the software list on the wooster.hut.fi site in Finland
and Johan's compenduim on the TAPR web site.

cu es 73 de Danie zs6awk

>Hi All,
>
>Let me just recap what the possibilities are:
>
>- A 60 MHz 56002, concervatively rated. Some have shown to run reliably at
>80 MHz.   (When executing multiple instructions per clock tick, this
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> to 60*3=180 MIPS or at a minimum, one instruction per clock tick - 60 MIPS).
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> fast as the 16-bit DSP chip as used in the new Pactor II modem.
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>- Comes with a very decent, full-screen, multiple windows, debugger that was
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>- A port of the Alef Null monitor "Leonid" bridges the EVM and the Finnish
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>Forgive me if I sound like a Motorola sales person - I don't work for them,
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>
>--Johan, KC7WW
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>>Mike N4CNW
>>
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>

66. Re: [HFSIG:1459] Re: EVM56002 ?

From: forrerj@peak.org (Johan Forrer.)
Date: Tue, 06 Aug 1996 14:58:31 -0800
To: hfsig@tapr.org
Message-ID: <199608062209.PAA00762@PEAK.ORG>
Rendered body type: text/plain

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> This is a 24-bit machine (120 dB dynamic range); for comparison, its
TWICE as
> fast as the 16-bit DSP chip as used in the new Pactor II modem.

Correction: MIPS factor is actually CLOCK/2, thus the 66 MHz chip
runs at 33 MIPS and the 80 MHz versions at 40 MIPS.

Sorry about that.

--Johan

>
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>>
>>Mike N4CNW
>>
>>
>
>

67. Re: [HFSIG:1440] Re: Amateur culture

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 6 Aug 1996 20:46:11 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608070346.UAA18246@servo.qualcomm.com>
Rendered body type: text/plain

>Well, what about its educational influence? I do not know whether this
>applies internationally, please advice me, but in Finland the amateur
>hobby has very often lead starting technical studies. In this way I
>think that society has

This is exactly my point. In my opinion, the climate toward technical education and experimentation in at least the US amateur world has been rather hostile, as evidenced by the attitudes toward the no-code license and spread spectrum and the tiny fraction of hams doing serious technical experimentation. Not to mention the fact that amateur technology is seriously behind the state of the art. And then you have the armchair lawyers that always seem to appear out of nowhere to claim that anything even slightly new and novel has got to violate some FCC rule somewhere, usually (but not always) the prohibition against codes and ciphers.

Ragchewing, contesting and DXing seem to be 99.9999% of ham radio. While I don't really object to these, it's hard to argue that they carry much weight with the non-amateur public that owns our spectrum.

The points you mention about experimentation, the training of operators and the like are specifically listed in the US amateur rules. Actually, I do agree that ham radio serves a valuable purpose even if the technical experimentation isn't leading-edge; just advancing one's own personal skills are enough. But even with this low standard, only a minority of the amateur population really take advantage of the opportunity. The rest engage in what can best be described as electronic stamp-collecting.

Phil

68. Re: [HFSIG:1460] Re: Probability question

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 6 Aug 1996 21:33:19 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608070433.VAA18353@servo.qualcomm.com>
Rendered body type: text/plain

Tom McD's answer:

>the probability of Y& words in a row are incorrect should be:

>   (1 - P^Y)^(S+1-Y)   (1)
[...]

I get a somewhat different answer. Or at least I followed a different analysis.

The probability that the very first Y symbols in the block of S (with S >= Y) are correct (with individual probability of symbol error P) is

        (1-P)^Y

Now add to this the probability that the first symbol is in error, but the next Y consecutive symbols are correct:

        P * (1-P)^Y

And the probability that the first TWO symbols are in error, but the next Y are correct (assuming S >=Y+2):

        P^2 *(1-P)^Y

and so on, summing each term until your "sliding window" of Y contiguous correct symbols bumps into the end of the block of S symbols. The resulting sum will be the overall probability of success.

This same analysis would have been directly applicable to another interesting amateur problem from years back. If your probability of error in copying a Morse character under exam conditions is a constant P, your errors are all independent (both obviously questionable assumptions) and the FCC rules required you to copy one solid minute out of 5, then what was your overall probability of passing the code test? :-)

Phil

69. Re: [HFSIG:1461] Pulses carrier propagation tests

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 6 Aug 1996 22:36:14 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608070536.WAA18424@servo.qualcomm.com>

Rendered body type: text/plain

Robert,

Very good work -- even if the results aren't as positive as I would have liked. I was especially interested to see you try different pulse durations and modulating waveforms. Since this is inherently a subjective test (i.e., is the signal "audible" or "annoying" to the SSB listener), giving subjective results is reasonable.

I concede that you've shown that under your specific test conditions, FHSS interference can be subjectively annoying to a human listener, even if it doesn't necessarily interfere with communications. This is a useful result. It means we need some form of dynamic coordination with ones' neighbors if SS is to be used operationally on the ham bands under similar conditions. I believe this is a practical idea, and that a flat prohibition of SS in the rules is still not necessary.

Indeed, it's not too hard to conceive of a scheme whereby a local narrowband user broadcasts his listen frequency on a VHF packet channel, alerting a nearby SS transmitter to "cut a hole" in his emission. Or it could be much simpler than that, by simply monitoring for strong local narrowband signals at the SS receiver and temporarily cutting them out of the hopping sequence for some period of time, such as a few minutes. I.e., the narrowband user need only transmit on the channel when the SS station is listening and that would do the trick. (I do confess this wouldn't necessarily work for split-frequency DX operations -- which brings us back to "protected DX subbands".)

Anyway, think of this adaptive channel-blanking scheme as a variation of my MACA multiple access algorithm, which is in turn based on "busy tone multiple access" (BTMA). The common theme to all three is the ability of a receiver to say "shhh, I'm listening" to its neighboring transmitters.

If the total number of hopping channels is large compared to the number of blanked channels, the SS transmitter probably wouldn't even have to tell the receiver about the new hopping sequence. FEC coding would fix the gap at the receiver.

Anyway, there are all sorts of clever ways one might mitigate SS/narrowband interference; I came up with this one in just the few minutes that I've been typing this message. If we succeed, I think the FCC would look quite favorably on us as they've been wrestling with these sorts of dynamic spectrum sharing issues for a long time. The ham bands are a perfect testing ground for this kind of thing. And I can think of other applications for adaptive channel protection -- operating in radio quiet zones, for example. Instead of having to stay off the air entirely within the zone, you'd monitor a special packet channel and forego transmitting only when the protected receiver is actually in operation.

My one major quibble with your results comes from your 30dB extrapolation from the whip RX antenna to a "good amateur station". I would like to see you try it with a *real* "good amateur station". I.e., a local ham friend, if one is close enough to you, especially if he/she has a beam that can be rotated to test the effects of antenna directivity.

In the further work category, it would be interesting to try some objective tests of simulated FHSS interference against existing amateur digital schemes. It would be especially interesting to see how Clover, Pactor, G-TOR etc compare to each other against this kind of pulsed QRM. If the FEC in these schemes are able to handle it well, then one possibility is to permit spread spectrum within those band segments where these schemes are currently popular. Of course, being smaller than a whole band the "hit rate" on any given channel would be greater. But with the increasing popularity of these schemes, perhaps the hopping range will soon be extended to the whole "CW" portion of each band. :-)

Hmm. Perhaps instead of having band segments for "DX" and "local" (or perhaps in addition), we could divide up the HF bands into "machine" and "manual" subbands. Existing CW and SSB/AM would be combined into a "manual" subband, while the "machine" section would accomodate all digital schemes intended for machine reception, including spread spectrum. The thinking, of course, is that emissions in the "machine" subband would be likely FEC coded in a way to resist the burst interference of FHSS while the "manual" emissions would not.

And there is again the possibility of simply not hopping at all unless
you have to (e.g., when the band is evenly crowded). But this leaves
us with another question -- given the clear benefits of FEC coding,
which substantially reduces power requirements at the expense of
bandwidth, how much wider should such signals be allowed to be
assuming they don't take up the whole band?

One approach to the problem is to count how many nearby neighbors N
you are likely to have on the band at any time and let each use no
more than 1/N of the band. That way you can preserve just enough
orthogonality to solve the near-far problem without denying everyone
the benefits of at least some bandwidth expansion. This is really just
an explicit, manual form of the "adaptive notching" mentioned above,
and I tend to prefer automatic methods over manual ones.

Comments?

Phil

70. Re: [HFSIG:1446] Re: Amateur culture

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 7 Aug 1996 01:27:48 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608070827.BAA18788@servo.qualcomm.com>
Rendered body type: text/plain

>As it applies to the current HFSIG discussions, I think that part of
>our educational process should include investigating new techniques and
>technologies. On that basis, I'd love to see some exploration of SS
>techniques on HF. But since nowhere in Part 97 is the rationale for
>Amateur Radio stated as the development of a maximally efficient
>communication system, it's not a given--at least to me--that we should
>abandon all narrowband activity in favor of an SS system even if such
>a system can be shown to be more spectrum-efficient in theory and
>practice. Which means that any SS system must coexist with narrowband
>uses of the spectrum by whatever means make that possible.

There are compelling reasons to investigate spread spectrum that go
beyond spectral efficiency -- as important as that may be, both to us
directly and as a testing ground for other services. SS is
increasingly important as a commercial technology, and this trend is
certain to continue. If the Part 97.1 notion of "training a reservoir
of skilled technicians" is to have any relevance to the outside world,
this is a technology that amateurs should be exposed to, preferably in
as wide a variety of settings (e.g., frequency bands) as
possible. This doesn't mean completely displacing traditional
narrowband operations, but it does mean that reasonable accomodations
should be made for it instead of banishing it entirely to the upper
microwave spectrum (the NIMBY syndrome).

Phil

71. RE: [HFSIG:1454] EVM56002 ?

From: Stephan Sykes <ssykes@emirates.net.ae>
Date: Wed, 7 Aug 1996 06:46:15 +0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BB847C.8A8E18C0@csa106.emirates.net.ae>
Rendered body type: text/plain

If it is around the price you quoted, please count me in.

Steve Sykes KD2OM/A61AA

From: Greg Jones[SMTP:wd5ivd@tapr.org]
Sent: Tuesday, August 06, 1996 8:26 AM
To: hfsig@tapr.org
Subject: [HFSIG:1454] EVM56002 ?

There is an opportunity for TAPR to do a group purchase on the Motorola
EVM56002 at a pretty low cost.   Between 20 and 200 units would be the
goal...same price no matter how many are purchased. Figure about $60 under
current commercial prices. No hard prices, becuase more work would have to
be done.

Would there be interest in this in the HF SIG ? Espiecally since Johan's
code would run on it.

I have spoken with Johan and he thinks it would be great...what are some

other reactions.

Cheers - Greg, WD5IVD

72. Re: [HFSIG:1432] Re: Wide band modes

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Tue, 6 Aug 1996 11:08:32 -0400
To: hfsig@tapr.org
Message-ID: <208A69E0.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

>You are exactly right -- different convolutional codes can indeed be
>used to differentiate between signals; it's an entirely viable form
>of CDMA. If the code rate is low enough, then this is virtually
>identical to coded direct sequence spread spectrum! In fact, Andrew
>Viterbi invented such a scheme a few years ago and published a paper
>about it.

Phil,

What is the name of the paper you are talking about here?

Tony

73. Re: [HFSIG:1429] Re: SS on HF bands

From: LANIER.R.A-@smtpgty.bwi.wec.com (LANIER.R.A-)
Date: Tue, 6 Aug 1996 11:24:35 -0400
To: hfsig@tapr.org
Message-ID: <208A7380.1858@smtpgty.bwi.wec.com>
Rendered body type: text/plain

While experimentation and self-training are certainly important
reasons to keep ham radio going, they are no the ONLY reasons. Has
anyone seen the movie 'Independence Day' ? After the aliens had
basicly taken the world over, all satellite communications were
broken. World governments could not communicate with each other.
However, there was one way they could - amateur radio using morse
code! There is a scene where radio operators are franticly using keys
to relay information.

This was a very positive image for ham radio operators. Keeping
amateur radio going for BOTH emergency communications AND advancing
the state of the art is why amateur radio should be kept alive.

Ham radio is NOT just a hobby!!!

73s de
Tony, KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:1429] Re: SS on HF bands
Author: hfsig@tapr.org at BALT.SMTP
Date: 8/5/96 12:40 AM

I guess it boils down to what you think ham radio is all
about. Reliable communications? That description hardly fits direct
long-haul HF. Easily accessible to those without large lots, liberal
housing covenants and money for big antennas and power amplifiers? I
think not. The popularity of VHF/FM and repeater networks shows there
are a lot of hams who are interested primarily in reliable local
communications with small stations, and who aren't purists when it
comes to using relays. Spread spectrum promises to provide the same
capabilities over wider areas on HF.

I confess that perhaps my view of ham radio is somewhat different than
yours. About the only activity that has a hope of justifying our bands
for very long is technical self-training, experimentation and
advancement. That means keeping up with the state of the art in the
non-amateur world, and spread spectrum is certainly one technology
where we trail the commercial and military world. While I don't really
have anything against DXing and contesting, I really don't see how they
help justify ham radio's existence to the outside world; they're really
little more than technological stamp-collecting.

I don't want to do away with traditional hamming, but I don't want it
to choke experimentation either. While I can't guarantee that there

won't ever be any SS interference to narrowband modes, I think it can
be minimized by careful system design and operational coordination.
And whatever risks there are will be worth the benefits. After all,
ours is not a critical safety-of-life service like aviation; we're
supposed to be experimenters, and experiments can fail. That's part of
the process.

Phil

74. Re: [HFSIG:1464] Re: Amateur culture

From: Pete Hardie <hardie@duke.usask.ca>
Date: Wed, 7 Aug 1996 11:01:43 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <Pine.OSF.3.95.960807103956.23754A-100000@duke.usask.ca>
Rendered body type: text/plain

On Tue, 6 Aug 1996, Phil Karn wrote:

> Ragchewing, contesting and DXing seem to be 99.9999% of ham radio.
> While I don't really object to these, it's hard to argue that they
> carry much weight with the non-amateur public that owns our spectrum.
>
> ....
> [They] engage in what can best be described as electronic stamp-collecting.

Two points Phil:

1. Without those 99.9999%, the remaining .0001% would not have *any*
spectrum to experiment in at all.

2. The vast majority of amateurs (myself included) do not have the ability
or, to be generous, sufficient education to ever be in a position to
experiment with SS or other advanced digital modes of communication.

I think the idea that amateurs ever made any significant advances in radio
technology is largely a myth. There's no way that your average ham, 60 or
more years ago, could start with a CW/AM rig and tinker around with it and
come up with SSB. Any ham who did make an advance in technology was, like
Phil, a highly educated person who already had the background to be able
to understand what they were doing *without* a ham radio license. All the
license did for them was give them access to spectrum where they could
experiment.

73 de Pete (still struggling to make that EZ-KIT transmit a packet :-)
ve5va.qrp@usask.ca

75. Re: [HFSIG:1471] Re: Amateur culture

From: Charles Brain <chbrain@dircon.co.uk>
Date: Wed, 07 Aug 1996 22:11:15 +0100
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19960807211115.0067efd0@popmail.dircon.co.uk>
Rendered body type: text/plain

I would like to disagree with the following because if it had not been for
amateur radio I would never have got interested in electronics, never
would have done a degree in the subject and never would have earned
a living doing electronics.

My first introduction to the hobby was at the age of eleven when I used
to listen to old timers Rag chewing on 160 m.

Also an amateur in the early days was reported to have told Marconi
when he met him that he was 'only' an amateur, Marconi replied
so am I.

>I think the idea that amateurs ever made any significant advances in radio
>technology is largely a myth. There's no way that your average ham, 60 or
>more years ago, could start with a CW/AM rig and tinker around with it and
>come up with SSB. Any ham who did make an advance in technology was, like
>Phil, a highly educated person who already had the background to be able
>to understand what they were doing *without* a ham radio license. All the
>license did for them was give them access to spectrum where they could
>experiment.
>
>73 de Pete (still struggling to make that EZ-KIT transmit a packet :-)
>ve5va.qrp@usask.ca
>

Regards Charles

76. Re: [HFSIG:1471] Re: Amateur culture

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 7 Aug 1996 18:02:19 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608080102.SAA19732@servo.qualcomm.com>
Rendered body type: text/plain

>1. Without those 99.9999%, the remaining .0001% would not have *any*
>spectrum to experiment in at all.

Why do you say that? The US rules specifically mention
experimentation, training and development. Presumably that (and public
service and international goodwill) is why they gave us spectrum.
They don't mention ragchewing, DX and contesting, one way or the
other.

>2. The vast majority of amateurs (myself included) do not have the ability
>or, to be generous, sufficient education to ever be in a position to
>experiment with SS or other advanced digital modes of communication.

Experimentation can be at many levels. One does not have to build the
technology up from the ground to use it or at least experiment with it
and see how well it works. Airtouch just announced that they're going
to provide Powerband service (their name for Qualcomm CDMA) at the
Republican Convention next week here in San Diego. And if your average
Republican can use spread spectrum, *anyone* can. :-)

>I think the idea that amateurs ever made any significant advances in radio
>technology is largely a myth. There's no way that your average ham, 60 or

Well, it's certainly true that the vast majority of hams have made no
significant advances in radio. But I strongly disagree with your
apparently absolute statement. Yes, many of the advances I attribute
to ham radio were in fact made in the commercial realm by people who
happened to have ham licenses. But even then, ham radio still deserves
at least some of the credit.

In my own case, I got my first ham license at age 14. I credit it with
steering me toward my career. Since graduation, I've managed to
maintain a symbiotic relationship between my ham radio and work
activities. And I'm finding this is fairly common among my colleagues
who are also hams. One of my company's founders, Klein Gilhousen
(WT6G) is fond of saying that "anything you do in ham radio you'll
probably find a use for at work within 6 months." I think I've
demonstrated that several times already. Some examples:

I came up with an improved TCP round trip timing mechanism largely as
a result of my experience in running TCP/IP over amateur packet radio.
It's now a required part of the TCP spec.

I (re)invented what became Bellcore's S/KEY one-time password scheme one
night while visiting a fellow ham in NJ. I wanted to log back into my
UNIX system at home without typing my password in the clear on 2m.

My NOS software has become the foundation of quite a few commercial dialup
IP router packages. Etc.

There are other hams who could list similar accomplishments. Some of
Franklin Antonio's graphics work in Instant Track went into the early
Qualcomm Omnitracs dispatch displays. Quite a few of the RF designers
around here got their first exposure to this stuff through ham radio.
Those who developed Clover, Pactor and the like have seen their
systems used by non-amateur services. AMSAT's pioneering work in small
LEO satellites is now spawning a whole new industry (which,
ironically, is now threatening our spectrum :-(). And so on.

I think it's fair to say that this sort of stuff carries far more
weight with the FCC than does DXing, ragchewing and contesting, even
though only a small fraction of the amateur population is involved in
the former. And that's one reason why the FCC has actually been trying
to promote greater use of spread spectrum in the amateur service.

Phil

77. Re: [HFSIG:1472] Re: Amateur culture

From: lynnc@sos.net
Date: Wed, 7 Aug 1996 20:04:43 -0700
To: hfsig@tapr.org, hfsig@tapr.org

Message-ID: <199608080304.UAA04838@sos.sos.net>
Rendered body type: text/plain

Well, blood boiling now, not a good time to enter feelings, but by
gosh, we DID go from CW/AM to SSB and much, much more 60 (well, almost)
years ago. Formal training was ARRL license manual, ARRL handbook, and
perhaps for the more affluent, RADIO HANDBOOK.
     Still in high-school and with cheering older hams around, we did
create, innovate and build new and exciting stuff - - daily. We were
probably not even average, and probably re-invented most things because
technical knowledge didn't spread as fast and seamlessly as it does
today. I'm darn proud of my "average" ham friends of the late 1940's
and early 1950's---   W7KGV, W7LAN, W7LFA, W7FXD and hoards of others.
     I was pretty slow, and didn't come up with my first junk-box SSB
rig until 1960. Most of my friends had graduated to commercially built
SSB gear by then, and were into home-brew VHF stuff from the high
class surplus of the time.
          There is a place for everyone in this mess, we had better stick
togather more tightly than in the past, as the benevalent public is beginning
to look for more space, and the monitoring public would axe us in an instant
if they hear the stuff for which we are using our precious bands. (not that
the TV and cellular frequencies are a model to admire.)
                          but I digress------- Lynn, W7LTQ

78. Re: [HFSIG:1457] Re: EVM56002 ?

From: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Date: Wed, 07 Aug 1996 23:21:45 -0400
To: hfsig@tapr.org
Message-ID: <199608080321.XAA10065@appstate-01.campus.mci.net>
Rendered body type: text/plain

At 06:49 AM 8/6/96 -0500, you wrote:
>Greg Jones wrote:
>
>> There is an opportunity for TAPR to do a group purchase on the Motorola
>> EVM56002 at a pretty low cost.   Between 20 and 200 units would be the
>> goal...same price no matter how many are purchased. Figure about $60 under
>> current commercial prices. No hard prices, becuase more work would have to
>> be done.
>>
>> Would there be interest in this in the HF SIG ? Espiecally since Johan's
>> code would run on it.
>
>Sounds great considering the price is already only $149.95 at the
>Design-Net price page (http://www2.motorola-dsp.com/dsp/home/net/pg.html).
>
>They also offer a student discount of 20% reducing the price to $120 + $3
>shipping (http://www2.motorola-dsp.com/dsp/home/edu/discount.html)
>
>Mike N4CNW
>
>

If we are talking about less than $100 count me in!

73's
Don, K4JPJ

79. Re: [HFSIG:1469] Re: Wide band modes

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 7 Aug 1996 21:38:15 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608080438.VAA20197@servo.qualcomm.com>
Rendered body type: text/plain

>   What is the name of the paper you are talking about here?

"Very Low Rate Convolutional Codes for Maximum Theoretical Performance of
Spread-Spectrum Multiple-Access Channels", Andrew J. Viterbi, IEEE
Journal on Selected Areas in Communications, Vol 8 No 4, May 1990, p 641.

Abstract - This paper treats a spread-spectrum multiple-access
communication system for which both spreading and error control is
provided by binary PSK modulation with orthogonal convolutional
codes. Performance of spread spectrum multiple access by a large
number of users employing this type of coded modulation is determined
in the presence of background Gaussian noise. With this approach and
coordinated processing at a common receiver, it is shown that the

aggregate data rate of all simultaneous users can approach the Shannon
capacity of the Gaussian noise channel.

80. New RF safety standards

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 8 Aug 1996 01:52:18 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608080852.BAA20871@servo.qualcomm.com>
Rendered body type: text/plain

Some of you may have just heard that the FCC has adopted new
and more stringent rules regarding RF emission safety, and these
rules do NOT exempt amateurs as they have in the past.

See http://www.arrl.org/news/rfsafety for details.

It does occur to me that this might be the impetus the amateur service
needs to finally get serious about using the minimum power required to
maintain communications. It may even spark interest in newer, more
power-efficient modulation methods -- even if they need more power.

And who knows? We might even find our TVI/RFI problems reduced too.

Phil

81. Re: [HFSIG:1473] Re: Amateur culture

From: Mike Curtis <wd6ehr@kaiwan009.kaiwan.com>
Date: Thu, 8 Aug 1996 02:05:57 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608080905.CAA18844@kaiwan009.kaiwan.com>
Rendered body type: text/plain

>
> >1. Without those 99.9999%, the remaining .0001% would not have *any*
> >spectrum to experiment in at all.

> I think it's fair to say that this sort of stuff carries far more
> weight with the FCC than does DXing, ragchewing and contesting, even
> though only a small fraction of the amateur population is involved in
> the former. And that's one reason why the FCC has actually been trying
> to promote greater use of spread spectrum in the amateur service.

One big problem I see with amateur radio is that the cart has been put
before the horse. The "99.9999%" has made it quite difficult for the
innovators to experiment. Yet innovation is a major portion of the
cornerstone of amateur radio. In areas like Los Angeles, it is doggone
nearly impossible to experiment on popular bands such as 2 meters and open
HF bands. 220 is better, but only because laws were changed that MANDATED
setting aside spectrum for "weak signal" stuff. We have old, established
technologies putting ironclad claims on most of the spectrum, and this is
IMHO exacerbated by actually encouraging prospective hams to memorize by
rote the test answers. Yes I know we live in the age of instant
gratification. I just think it's inherently evil.

 -- mike

82. Re: [HFSIG:1448] How much QRM is too much?

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 8 Aug 1996 02:30:47 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608080930.CAA20982@servo.qualcomm.com>
Rendered body type: text/plain

>For someone trying to copy a signal right at the thermal, cosmic, or
>atmospheric noise level, any interference is appreciable. For some hams

"Any" is a rather absolute term; it reminds me of the absolutist
position on biological radiation exposure (no artificial radiation
whatsoever is tolerable even though the natural background can vary
substantially). But unlike that particular can of worms, the math
here is not too hard.

I've been meaning to work out the exact figures for how long an
observing interval is required to even detect a certain small change
in the noise floor with a certain confidence. Tom Clark (W3IWI) does

essentially this kind of calculation all the time in radio astronomy, where all the signals he observes are essentially noise (at least until the first positive SETI acquisition). He sketched out the basic principles for me over lunch a while ago. Once I translated it from "radio astronomerese" into "communicationese" I realized it was all the same math I'm used to seeing in comm theory.

Qualitatively, the minimum required observing interval depends on both the difference in noise level (larger differences can be detected more quickly) and the observation bandwidth (detection is quicker with larger observation bandwidths -- so the obsession with narrowband receivers works to my advantage here! :-))

Phil

83. [HFSIG:1473] Re: Amateur culture

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 8 Aug 96 11:13:03 +0100
To: hfsig@tapr.org
Message-ID: <H000029202318a16@MHS>
Rendered body type: text/plain

Hi Phil,

> I think it's fair to say that this sort of stuff carries far more
> weight with the FCC than does DXing, ragchewing and contesting, even
> though only a small fraction of the amateur population is involved in
> the former.

Yip, I agree, but the 'DXing, ragchewing and contesting' and the likes is where it all starts, and is what keeps it alive in the general population. If amateur radio was only the 0.0001% expert experimenters I would have probably never heard of it, and neither would most of the 0.0001%.

'Technological stamp collecting' is in my opinion an essential part of amateur radio, not to be lightly disregarded.

> And that's one reason why the FCC has actually been trying
> to promote greater use of spread spectrum in the amateur service.

I would hope the FCC is doing this with a wider view of amateur radio, or does it intend to promote the amateur bands as a test bed for commercial experimentation, rather than for more personal radio experiments? Is this how it is justifying an 'amateur' allocation in a commercial enviroment?

This issue caused a bit of controvercy in New Zealand when a number of hams with commercial interests started using ham bands for commercial experimenation, although not blatently. Others saw it as a usefull contribution to amateur radio. I wouldn't like to say, it sure is on the fine line between 'advanceing the art' and 'pecuniary interest'.

Certianly, that line is oversteped when 'pecuniary interest' takes presidence over other legitimate uses of the bands. (thinking more of the FCC here)

Rob

84. Re: [HFSIG:1480] Re: Amateur culture

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 8 Aug 1996 09:07:37 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608081407.JAA22454@www.kelly-afb.org>
Rendered body type: text/plain

Rob,

In your message you write:
[Stuff Deleted]
>
> This issue caused a bit of controvercy in New Zealand when a number of
> hams with commercial interests started using ham bands for commercial
> experimenation, although not blatently. Others saw it as a usefull
> contribution to amateur radio. I wouldn't like to say, it sure is on the
> fine line between 'advanceing the art' and 'pecuniary interest'.
>
> Certianly, that line is oversteped when 'pecuniary interest' takes

> presidence over other legitimate uses of the bands. (thinking more of
> the FCC here)
>
>
> Rob
>
>

        In past years, educational institutions used hambands extensively
        for education/training. Many high schools had ham radio as in part
        of the "shop" program and most electronic technical schools also
        used ham radio as a teaching tool. While I don't think that
        colleges and/or universities used ham radio as a tool at the under-
        graduate level, I know that is was used in lab work for post-
        graduate degrees. And I have to admit that I believe in most cases
        high schools, technical schools and colleges/universities were not
        too concerned whether or not a student had a ham radio license as
        long as there was a ham present in the "lab".

        Also, I know of three specific instances where the FCC has granted
        licenses to commecial businesses to operate in the hambands...granted
        there were frequency restrictions, and special licenses...even
        restrictions on who the stations could talk with...in some instances,
        they were issued special callsigns, other times regular ham callsigns.
        These stations were licensed to do experimental and/or production unit
        field test.

        What's new?

Walt/K5YFW

| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development. |

	The greatest dangers to liberty
Walt DuBose - K5YFW	lurk in insidious encroachment
E-Mail k5yfw@www.kelly-afb.org	by men of zeal, well-meaning
Business Telephone: (210)925-6081	but without understanding.
Home Telephone: (210)696-3196	
	- Justice Louis D. Brandeis

85. Re: [HFSIG:1478] Re: Amateur culture

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 8 Aug 1996 09:11:10 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608081411.JAA22482@www.kelly-afb.org>
Rendered body type: text/plain

In your message you write:
[Stuff Deleted]
>
> One big problem I see with amateur radio is that the cart has been put
> before the horse.

        The horse has got to go...the horse is old technology and the
        droppings are causing a mal-odor in our hobby.   -- Walt/K5YFW

86. [HFSIG:1481] Re: Amateur culture

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 8 Aug 96 16:23:14 +0100
To: hfsig@tapr.org, k5yfw@www.kelly-afb.org
Message-ID: <H000029202323446@MHS>
Rendered body type: text/plain

> In past years, educational institutions used hambands extensively
> for education/training. Many high schools had ham radio as in part

Granted, and I support such activities, such as JOTA etc. This is well
in the spirit of amateur radio.

> Also, I know of three specific instances where the FCC has granted
> licenses to commecial businesses to operate in the hambands...granted
> there were frequency restrictions, and special licenses...even

> restrictions on who the stations could talk with...in some instances,
> they were issued special callsigns, other times regular ham
> callsigns. These stations were licensed to do experimental and/or
> production unit field test.

I'm sure there are cases where such commercial use is justified, and I never said I opposed the semi-commercial experimentation in New Zealand (they did not have special commercial licences), I'm just wondering about the fine line between commercialisation of the ham bands and the advancement of technology. International treaties specifically exclude 'pecuniary interest' in amateur radio. I'm wondering if perhaps the FCC are promoting commercial interests *over_and_above* the interests of all amateurs, and possibly against the International Radio Regulations. Perhaps the FCC see the amateur bands as more for experimentation for the advancement of commercial technology, rather than for the self training of amateurs as specified in the international regulations.

Commercial enterprises can still experiment on other bands with other licences, but they cost more money than the ham bands. The non-commercial nature of amateur radio works too ways, to protect industry from amateurs exploiting their allocations commercially, and to protect amateurs from commercial organisations trying to exploit a cheap resource to the detriment of amateurs, and their self training.

I was wonder if this is why they appear to support spread spectrum (an emerging commercial technology) on HF, apparently disregarding the interests of the majority of amateurs.

Rob

87. New floating point EVM from TI

From: forrerj@peak.org (Johan Forrer.)
Date: Thu, 08 Aug 1996 10:18:45 -0800
To: HFSIG@tapr.org
Message-ID: <199608081729.KAA23715@PEAK.ORG>
Rendered body type: text/plain

Hi All,

Those interested in dabbling with DSP hardware; TI's new 320C31 DSK is finally available. Marshall (1-800-522-0084) has them at a special introductory price of $79.

This is a FLOATING POINT DSP chip. I'm not sure about the clock speed, CODEC, or how much memory the DSK has, but I'll let you know when I receive mine.

I have a small collection of software tools for this DSP and am particularly interested in the floating point architecture for implementing FFT demodulators where the extra dynamic range is essential.

Exciting times - there also is some other interesting developments from Motorola that I'll tell you more about in the near future.

--Johan, KC7WW

88. RE: EVM56002 ?

From: rhiii@pop.erols.com (Richard H)
Date: Thu, 08 Aug 1996 18:17:00 GMT
To: hfsig@tapr.org
Message-ID: <320a2920.44799027@smtp.erols.com>
Rendered body type: text/plain

Question to those familiar with this kit.

What is the knowledge entry level for this DSP kit ? Does it assume the student already be fluent in FFT, Z and LaPlace transforms. Can this be considered a "First" DSP learning kit or is it targeted towards the practicing dsp engineer ?

Thanks,
de Richard/N2JR

89. Re: [HFSIG:1480] Re: Amateur culture

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 8 Aug 1996 12:31:19 -0700 (PDT)

To: hfsig@tapr.org
Message-ID: <199608081931.MAA14708@servo.qualcomm.com>
Rendered body type: text/plain

>I would hope the FCC is doing this with a wider view of amateur radio,
>or does it intend to promote the amateur bands as a test bed for
>commercial experimentation, rather than for more personal radio
>experiments? Is this how it is justifying an 'amateur' allocation in a
>commercial enviroment?

Amateur radio has never existed in a vacuum. Although direct
commercial exploitation is not allowed, it has always been one of the
goals of the amateur service to develop technology and to train
engineers and radio operators who can then benefit commercial and
government activities. It's a win-win situation. Not only does that
create new technology for our use, but it earns lots of brownie points
for ham radio.

I believe "commercial exploitation" should be narrowly defined as
selling a public communication service for profit. Developing new
technology on the ham bands that may then be commercialized is in
itself perfectly legitimate, although I am concerned by the recent
trend towards keeping such technology proprietary.

Phil

90. Re: [HFSIG:1482] Re: Amateur culture

From: mwestfal@csci.csusb.edu (Michael Westfall)
Date: Thu, 8 Aug 1996 16:46:35 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608082346.QAA22063@csci.csusb.edu>
Rendered body type: text/plain

> > One big problem I see with amateur radio is that the cart has been put
> > before the horse.
>
>
>   The horse has got to go...the horse is old technology and the
>   droppings are causing a mal-odor in our hobby.   -- Walt/K5YFW

"The warp-drive thrusters have been put before the starship" ???
>
>

--

---
73 de Mike "They call me `MiKUY'", N6KUY
                ax.25net: N6KUY@W6JBT.#SOCA.CA.USA.NOAM
                internet: mwestfal@.csci.csusb.edu
                     web: http://web.csusb.edu/public/csci/mwestfal
Linux: the Gates of Hell shall not prevail.
---

91. Re: [HFSIG:1485] RE: EVM56002 ?

From: forrerj@peak.org (Johan Forrer.)
Date: Thu, 08 Aug 1996 19:35:18 -0800
To: hfsig@tapr.org
Message-ID: <199608090246.TAA26420@PEAK.ORG>
Rendered body type: text/plain

Richard,

>Question to those familiar with this kit.
>
>What is the knowledge entry level for this DSP kit ?   Does it assume
>the student already be fluent in FFT, Z and LaPlace transforms.

Not quite a requirement, but that surely comes in handy. Without that, you
should consider joining a support group where you can share the experiences
of others. In time you will broaden you horizon and the formiddable math

will become a lot of fun.
Since you already know the buzz words, I would say that you are well on your
way.

>Can this be considered a "First" DSP learning kit or is it targeted towards
>the practicing dsp engineer ?
>
>Thanks,

>de Richard/N2JR
>
>

There are of course ways to learn about DSP without having DSP hardware,
i.e., PC program packages. However, if you are in a position where you would
like to use DSP hardware in your experiments or your career track, the EVM's
are often the key to getting started. That goes for engineers or scientific
minds. It does take a combination of some skill, level of interest,
persistence, and intuition to tame the little beasty.

Hope this helps.

--Johan

92. Re: [HFSIG:1487] Re: Amateur culture

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Fri, 9 Aug 1996 07:49:17 -0500 (CDT)
To: hfsig@tapr.org
Cc: karn@qualcomm.com, choffman@pelican.davlin.net (Ric Hoffman),
forrerj@frl.orst.edu (Johan Forrier)
Message-ID: <199608091249.HAA03844@www.kelly-afb.org>
Rendered body type: text/plain

Mike,

In your message you write:
>
> > > One big problem I see with amateur radio is that the cart has been put
> > > before the horse.
> >
> >
> >   The horse has got to go...the horse is old technology and the
> >   droppings are causing a mal-odor in our hobby.   -- Walt/K5YFW
>
>
> "The warp-drive thrusters have been put before the starship" ???
> >
> >

        That's a perfectly acceptable practice as long as you
        SPREAD THEM.   -- Walt/K5YFW

>
>
> --
>
> --
> 73 de Mike "They call me `MiKUY'", N6KUY
>   ax.25net: N6KUY@W6JBT.#SOCA.CA.USA.NOAM
>   internet: mwestfal@.csci.csusb.edu
>   web: http://web.csusb.edu/public/csci/mwestfal
> Linux: the Gates of Hell shall not prevail.
> --
>
>

--
```
=========================================================================
| The MicroSoft operating system didn't get as bad as it is overnight,|
| it has taken over 10 years of careful, calculated development.      |
=========================================================================
|                                 |                                     |
|                                 | The greatest dangers to liberty     |
| Walt DuBose - K5YFW             | lurk in insidious encroachment      |
| E-Mail k5yfw@www.kelly-afb.org  | by men of zeal, well-meaning        |
| Business Telephone: (210)925-6081 | but without understanding.        |
|     Home Telephone: (210)696-3196 |                                   |
```

```
|                                 |   - Justice Louis D. Brandeis   |
|                                 |                                 |
==========================================================================
```

93. new C31 EVM ?

From: zs6awk@global.co.za (Danie Brynard)
Date: Fri, 09 Aug 1996 17:07:15 +0200
To: hfsig@tapr.org
Message-ID: <199608091904.RAA07706@lin01.global.co.za>
Rendered body type: text/plain

Did anyone play with the new TMS320C31 EVM yet ? How usefull is it for ham radio datacomms experiemnts ? Can one 'flash' applications ? Async Serial Ports ?

Danie
zs6awk@global.co.za

94. ECC for SHORT data block

From: forrerj@peak.org (Johan Forrer.)
Date: Fri, 09 Aug 1996 09:22:46 -0800
To: HFSIG@tapr.org
Message-ID: <199608091633.JAA20720@PEAK.ORG>
Rendered body type: text/plain

Hi Experts,

I wonder what a suitable ECC code would be for a short bit field such as 32-bits.
It need to detect and correct at least six bits. Further, need to have a fast algorithm to compute syndromes and error locations, if it could be done using a table lookup, that would be great.

The field and ECC code is for the header field following the sync vector in my new scheme. I was thinking of using a BCH code.

Any further suggestions?

--Johan, KC7WW

95. FTP mail archive?

From: Robert.Glassey@nmp.nokia.com
Date: Fri, 9 Aug 96 17:52:07 +0100
To: hfsig@tapr.org
Message-ID: <H00002920234821f@MHS>
Rendered body type: text/plain

Hi

I was wondering what has happended to the mail archive on the TAPR FTP site? It seems to have stopped in June.

Cheers,

Rob

96. Re: [HFSIG:1488] RE: EVM56002 ?

From: Richard Harrison <rhiii@pop.erols.com>
Date: Fri, 09 Aug 1996 07:31:20 -0700
To: hfsig@tapr.org
Message-ID: <320B4BB8.6A67@pop.erols.com>
Rendered body type: text/plain

Thank you Johan for taking the time to respond. Taking all of your comments under advisement, will order the EVM56002 and begin the journey towards understanding these new technical building blocks.

I am an EE but it has been quite a while since I've had to work with integro-differential equations and their associated transforms.

Regards,

de Richard/N2JR

97. Free Near-Field Analysis Software

From: k6sti@n2.net (Brian Beezley)
Date: Fri, 9 Aug 1996 12:52:18 -0700
To: cq-contest@tgv.com, moon-net@vm.stlawu.edu, hfsig@tapr.org,
wf1b-rtty@ve7tcp@ampr.org
Message-ID: <199608091952.MAA11031@ravel.n2.net>
Rendered body type: text/plain

As a result of the recent FCC ruling that mandates RF-exposure limits for amateur stations beginning January 1, 1997, I'm making available at no cost a special version of AO Antenna Optimizer software that calculates electric and magnetic near fields. NF.EXE requires a 386 or better, math coprocessor, VGA, and DOS 3.0 or later.

You can download the 245K NF.ZIP file from ftp://n6nd.nosc.mil. You may copy this free software for others as long as no charge is involved and the software is used for amateur purposes only.

After you unzip the file, see READ.ME for more information. Please carefully read the section on accuracy limitations of near-field modeling.

I hope this software helps hams evaluate their stations for compliance with the new FCC rule. The software should be especially useful at high-power contest, DX, and EME stations.

I'm providing this free software without support. The package includes extensive documentation and 92 example antenna files. I hope you'll refrain from calling, writing, or e-mailing questions about downloading or using the software. Thanks!

Brian Beezley, K6STI
k6sti@n2.net

98. FTP Packet Driver Specification

From: forrerj@peak.org (Johan Forrer.)
Date: Mon, 12 Aug 1996 20:17:06 -0800
To: HFSIG@tapr.org
Message-ID: <199608130327.UAA28713@PEAK.ORG>
Rendered body type: text/plain

Hi,

I would be grateful if someone could point me to the location of the "FTP packet driver specification". I have a suspicion that it may be somewhere at Clarkson, but was not able to locate it.

This is to be used to connect a new HF DSP modem a TCP/IP environment. I do have most of the low-level code sorted out, but miss the documentation to understand what is needed to interface to the network OS.

Thanks much.

--Johan, KC7WW

99. SUMMARY- *at least* Y *in a row* probability question

From: Richard Lay <lay@cod.nosc.mil>
Date: Tue, 13 Aug 1996 06:47:15 -0700
Cc: hfsig@tapr.org, lay@cod.nosc.mil
Message-ID: <32108763.7EDA@cod.nosc.mil>
Rendered body type: text/plain

Thanks to all that helped me with my probability question. I know much more about the problem now. Special thanks to Tim Firman, who spent much time on the problem and answering my questions. I also thank Jim Landis, whose solution appears first below. His is the one I implemented both in C (recursive and non-recursive forms) and MATHCAD (non-recursive only). If anyone would like a copy of either of these please let me know. And finally, Ken Butler went at the problem from a different point of view. I have only just received his solution, but it looks interesting, and I post it as it approaches the problem from another prospective. I received many more replies besides these. Thanks to all who responded. You are the ones who make the internet such a valuable resource.

Regards,
Rich Lay

****
Original question:
If the probability of an individual event occurring (heads on a coin, 3 on a die, an error in a received ALE word) is P, then within a sequence

of S events (S coin flips/die rolls/ALE words received), what is the
probability that *at least* Y of these events will occur *in a row* ?
****
Jim Landis solution:
I have a solution that is a little heavy in computation, but it does
solve the problem exactly, with out double counting or miscounting
any cases.

I don't know of any simple solution to that problem in "closed"
notation, but it yields very easily to a recursive formula, i.e. a
formula defined in terms of itself.

Define f(n, Y, S, p) as follows :
f(0, Y, S, p) = 1
f(n, Y,S, p) = 0   for all n > S >= 0
f(n, Y, S, p) = p*f(n-1, Y, S-1,p) + (1-p)*f(Y, Y, S-1, p)

n represents the number of successes needed to complete the current

        "streak", and should initially be set to Y
Y represents the total number of successes needed in a row
S is the number of remaining trials
p is the probability of success on any trial

f() could be interpreted as "the probability that you will have a
streak of n successes immediately, or a streak of Y successes later
on". The two terms evaluate the remaining probability after each roll
of the die. First term corresponds to success on the current roll, the
second term to failure.

So the probability of an unbroken string of four 3's in 20 rolls of a
fair die is f(4,4,20, 1/6)   or approximately 0.011026 .

You can verify that Y=1, S=1 yields a probability p, as you would
expect. You can also verify that Y=S yields p^^S.

This can be implemented in a simple computer program. However, because

of the double recursion, the run time of the program is O(2^^(S-Y)).
The formula is essentially called once for each of the 2 to the power
of S outcomes. For S=20, that is more than a million times. Beyond
S = 30 or so, that approach is so slow as to be impractical.

A non-recursive program could be written based on an S +1 by Y+1
array of values, and running in time O((S*Y)^^2). The approach would
be to create an array. Initialize one column of the array to value 1,
based on the first rule. Initialize one half of the array to 0, as
per the second rule. Every remaining cell of the array can be
calculated from 2 other cells by the function definition.
****
Ken Butler solution:
Anyway, assuming that independence and constant probability are OK here,
my
line of thinking is the following: let N be the total number of words
sent
when Y consecutive words are received incorrectly for the first time. For
example, if Y=3, in the following sequence of words sent (C = correctly
received, I = incorrect):

I C I I C C I I I

N = 9, since 9 words have been sent by the time the 3rd consecutive I
appears. Relating this to your problem, if N>S, then the radios link up,
whereas if N <= S, the link is disestablished. So prob(N>S) is the
probability you want. (This takes care of the "at least" problem, because
once Y consecutive I's show up, we stop looking.) You also see that the
last
four (=Y+1) words sent are a C followed by K I's; this must be the way
the
process stops, because if we'd gotten any more consecutive I's, we would
have
stopped earlier, and if we'd gotten any fewer, we'd keep looking.)

To show what happens with Y=2:

N cannot be 0 or 1;
N = 2 if we get I I, whose probability is P^2.
N = 3 if we get C I I, with prob. (1-P)*P^2.

N = 4 if we get I C I I or C C I I, ie. the last 3 are C I I no matter
what
the first is. Probability is (1-P)*P^2 as for N=3.

Then, a pattern:

N = 5 if we survive the first 2 words and then get C I I;
N = 6 if   3   C I I;
..
..
N = m if   m-3   C I I,

so that prob(N = m) = prob(N > m-3) * (1-P) * P^2.

Overall, the probability is 0 for a bit, P^2 once, (1-P)*P^2 for a bit,
and
then the above relationship holds. This was for Y=2, but the same idea
generalizes to other Y:

Prob(N = m) = : 0 if m < Y-1
                 P^Y if m = Y
                 (1-P) * P^Y * prob(N > m-Y-1) if m > Y.

Because each probability depends on previous ones, you'll need to
calculate
these probabilities one at a time, starting from 0 (or Y) and continuing
up
to S. It's probably easiest to keep track of prob(N > m), since this is
the
quantity of interest anyway, noting that prob(N > m) = prob(N > m-1) -
prob(N = m), and prob(N > 0) = 0.
****

100. New modem

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 13 Aug 96 17:58:42 +0100
To: hfsig@tapr.org
Message-ID: <H00002920238df82@MHS>
Rendered body type: text/plain

> This is to be used to connect a new HF DSP modem a TCP/IP environment.
> I do have most of the low-level code sorted out, but miss the
> documentation to understand what is needed to interface to the network
> OS.

Hi Johan,

I wonder how you are getting on with your new modem. Perhaps you would
like to post a brief description of it to the group.

I gather it uses two independent tones with PSK, or is this a multi tone
OFDM or MFSK modem?

Cheers,

Rob

101. Re: [HFSIG:1497] New modem

From: forrerj@peak.org (Johan Forrer.)
Date: Tue, 13 Aug 1996 14:00:50 -0800
To: hfsig@tapr.org
Message-ID: <199608132111.OAA00166@PEAK.ORG>
Rendered body type: text/plain

Hi Rob,

>I wonder how you are getting on with your new modem. Perhaps you would
>like to post a brief description of it to the group.
>
>I gather it uses two independent tones with PSK, or is this a multi tone
>OFDM or MFSK modem?
>
>Cheers,
>
>Rob
>

>

Sorry that I have not posted more details - just not had much time to work on and refine a specification as yet. Hopefully more at the DCC.

A short(?) description for those that cannot wait :-)

As things stand at the moment, it is FDM (not OFDM);four RC-shaped DQPSK waveforms in parallel each at a 75 baud symbol rate - 600 Hz at -60 dB. Modulation is DQPSK that varies from BPSK to 16-DQPSK, selectable to match conditions. The full bandwidth is always used, even at low data rates, however, both time and frequency diversity is then employed. The idea is to use a common DQPSK demodulator that does'nt know the difference - it just demodulates the input waveform as if it were m-DQPSK. That is the job for the packet dissasembler that knows how to unpack the data from information provided in a data header. For that purpose, each packet contains a header that provides a) frequency tracking information, b) bit phasing/clock information (extracted from the sync vector), and c) packet formatting/status information (an ECC protected field). Actual data contents are convolutionally coded using a variable rate code on the inner level, with an outer level block code.

Three modes of operation is planned:

Mode A: Unconnected broadcast mode with full ECC.
Mode B: Hybrid ARQ.
Mode C: Networked (CSMA) with encapsulated TCP/IP.
Mode D: Read-along mode. It is possible for a third party to decode and
        read traffic, pretty much as we can do with packet at the moment.

All specifications will be published for the benefit of those that wish to implement the protocol. It will be an "open architecture" although I hope to encourage commercialization - consulting and user-support services can be arranged. One essential item that would be made available is a "golden" CD-ROM - this will contain the printed standard, also test and diagnostic signals that will help one to verify whether your implementation conforms to the standard or not.

Hope this helps in the interim,

--Johan, KC7WW

102. Sound Cards

From: Mike Kerry <100577.1452@CompuServe.COM>
Date: 14 Aug 96 09:23:49 EDT
To: "(unknown)" <HFSIG@tapr.org>
Message-ID: <960814132349_100577.1452_GHW94-1@CompuServe.COM>
Rendered body type: text/plain

Can anyone please recommend a good PC sound card
which also can be programmed for DSP applications?

(I think this means having the 21xx chip).

Thanks

Mike G4BMK

103. Re: [HFSIG:1499] Sound Cards

From: Johan Forre <forrerj@peak.org>
Date: Wed, 14 Aug 1996 08:47:00 -0700 (PDT)
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <Pine.SUN.3.91.960814084143.29776C-100000@kira>
Rendered body type: text/plain

Hi Mike,

There used to be a number of manufacturers that used an Analog Devices three-chip set (included the 2115 DSP) on their sound cards. Now they are very scarce. Look at the Echo Speech DSP I on their web site: www.echospeech.com I also have just gotten a catalog from Jameco and see the sell the Orchid Soundwave 32 for 129 bucks. That is a 20 Mhz version DSP and I have one that I'm very happy with.

The other alternative is to look for a Turtle Beach Tahiti card - that one though has a 56002. There is a software toolkit available for that one as well. It's a really nice PC plugin card.

Hope this helps.

--Johan

On Wed, 14 Aug 1996, Mike Kerry wrote:

> Can anyone please recommend a good PC sound card
> which also can be programmed for DSP applications?
>
> (I think this means having the 21xx chip).
>
> Thanks
>
> Mike G4BMK
>
>

104. Re: [HFSIG:1491] ECC for SHORT data block

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 14 Aug 1996 11:02:28 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608141802.LAA13843@servo.qualcomm.com>
Rendered body type: text/plain

>I wonder what a suitable ECC code would be for a short bit field such as
>32-bits.

Is that 32 data bits plus however much overhead is necessary?

Why use such a small block? Performance is always better for larger
blocks.

The more I think about the problems of a fading channel, the more I am
troubled by schemes that separately encode sync, header and
data. There's something to be said for a scheme that encodes and
interleaves a continuous bit stream and then layers a simple packet
framing scheme on top that doesn't have to deal with frequent errors
(e.g., HDLC). Then the packet header would enjoy as much protection
as a much larger data field.

One way is a convolutional interleaver followed by a convolutional
encoder. A Reed-Solomon code could be layered on top of this to
further steepen the waterfall curve if a sync word is inserted to mark
the beginning of each RS block. Viterbi decoder and deinterleaver sync
can be achieved by a brute-force search of the possible states looking
for a good decoder metric to pop out. This is practical because there
aren't that many states to search. This is similar to the framing
approach used in digital video broadcasting.

The only drawback I can see is the getting-started overhead on short
transmissions. But if you need a certain amount of time diversity on
a fading channel to span the required number of channel coherence
times, you don't really have a choice -- a short transmission could
get swallowed by a fade. I.e., there's a minimum transmission time
requirement.

Phil

105. [HFSIG:1486] Re: Amateur culture

From: Robert.Glassey@nmp.nokia.com
Date: Wed, 14 Aug 96 19:35:28 +0100
To: hfsig@tapr.org
Message-ID: <H000029202338282@MHS>
Rendered body type: text/plain

>>I would hope the FCC is doing this with a wider view of amateur radio,
>>or does it intend to promote the amateur bands as a test bed for
>>commercial experimentation, rather than for more personal radio
>>experiments? Is this how it is justifying an 'amateur' allocation in a
>>commercial enviroment?
>
> Amateur radio has never existed in a vacuum. Although direct
> commercial exploitation is not allowed, it has always been one of the
> goals of the amateur service to develop technology and to train
> engineers and radio operators who can then benefit commercial and
> government activities. It's a win-win situation. Not only does that
> create new technology for our use, but it earns lots of brownie points

> for ham radio.

It's wider than that but yes, thats part of it.

> Developing new technology on the ham bands that may then be
> commercialized is in itself perfectly legitimate

Mostly, but it can get a little blured when the amateurs concerned already have a commercial application and simply use amateur radio as a vehicle to do commercial research. It's better when amateur research finds an application in the commercial world, where the amateurs concerned have no direct commercial interest, but it's very fuzzy.

> I believe "commercial exploitation" should be narrowly defined as
> selling a public communication service for profit.

The international radio regulations defines the amateur service in article S1.56 as:

"A radiocommunication service for the purpose of self training, intercommunication and technical investigations carried out by amateurs, that is, by duly authorised persons interested in radio technique soley with a personal aim and without pecuniary interest."

This definition is not on the adgenda of any upcomming WRC.

Nowhere in the regulations does it mention advancement of the state of the art. The only place it mentions the state of the art is when it requires that 'emmited frequency shall be as stable and free from spurious emmisions as the state of technical development for such stations permits'. Advanceing the art appears to be an addition of the FCC, which of course has no duristiction outside the USA.

Still, advancing the art is one motive for interest in radio technique (and the purpose of HFSIG) and should be encouraged, although never to the detriment of other equaly valid reasons for interest in radio technique. Of course it is usually entirly in the interests of amateurs in general, but the choice must be theirs, it should never be forced upon them especially in the name of some claimed benifit to a few.

Strictly speaking even the likes of CLOVER and PACTOR should have been developed with commercial expreimental licences, then the final tested products sold to amateurs. But this is just as tricky, the line is definatly blured. It's made for dificult when PACTOR II for example is sold to non-amateurs for high prices as a propitary commercial protocol.

The difficulity of course, is the risk of loosing the amateur element of amateur radio to commercial interests, and thereby either effectivly or actually loosing amateur bands, or even the entire service, as it may be seen as simply a commercial service and sold accordingly, with many eager buyers, and an equally eager seller.

Cheers,

Rob

106. Re: [HFSIG:1501] Re: ECC for SHORT data block

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 14 Aug 1996 15:05:55 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608142205.PAA17918@servo.qualcomm.com>
Rendered body type: text/plain

While waiting for a doctor's appt, I sketched out an approach to an extremely fade-resistant coding scheme along the lines I discussed earlier today.

The channel modulation can be any noncoherent scheme, such as DBPSK or M-ary FSK. The channel symbol stream is fed from a convolutional interleaver. Feeding the convolutional interleaver is a convolutional encoder of some rate, 1/N.

The interleaving degree is one more than some integer multiple of N, chosen to be large enough to give an adequate interleaving span for the symbol rate and channel coherence time. The extra input to the interleaver multiplexer is fed with a PN generator used for acquisition and synchronization.

The PN generator period corresponds to one "block length", and the

PN epoch is aligned so that it marks the beginning of the block. The
PN generator runs continuously throughout a transmission.

The FEC "block" could correspond to a fixed-size block of a
sequentially decoded convolutional code, or it could correspond to a
Reed-Solomon block that's layered on top of a Viterbi-decoded
convolutional code operating in stream mode, possibly with an extra
layer of interleaving between the RS and Viterbi coders to spread out
the error bursts from the Viterbi decoder. (This is a standard feature
of such concatenated codes.)

To acquire this signal, you DBPSK (or M-ary FSK) demodulate the
incoming symbol stream at all possible symbol timing phases and then
run a PN correlator with sampling taps spaced according to the
periodic appearance of the known PN sync sequence in the modulated
stream. When the correlator reports a sufficiently high peak, you now
have modem symbol timing, Viterbi decoder phase and Reed-Solomon block
phase. Or in the case of sequential decoding, you know where to begin
sequential decoding.

Alternatively, the "sync channel" input to the interleaver could be
all 1's, and the multiplexed output could be PN scrambled. The
advantage of this approach is that it is obviously quite easily
extended to PN scrambling at a faster rate, i.e., direct sequence
spread spectrum.

The result of all this effort is a series of decoded fixed-size data
blocks. Each block is almost certain to be either perfect or erased,
i.e., there should be few if any undetected errors at this level. Now
you can layer a conventional framing scheme, e.g., bit-synchronous or
octet-framed HDLC and carry a series of variable length packets.

This scheme seems to provide the maximum robustness possible against
fading. Because packets are all framed at a higher layer, no part of the
channel stream is more vulnerable than any other.

The biggest drawback is the fairly high overhead required to fill the
interleaver buffers at the start of transmission and to flush them at
the end, especially if a high degree of interleaving is required (as
it would be when the the symbol rate is relatively high and the
channel coherence time is relatively long). But some overhead is
inevitable if you really want protection against fading. (Consider a
standalone packet small enough to get lost in a relatively slow fade.)
Some tradeoffs are probably required here, as at some point you have
to fall back on retransmission (as when fading is very slow).

Phil

107. Re: [HFSIG:1501] Re: ECC for SHORT data block

From: forrerj@peak.org (Johan Forrer.)
Date: Wed, 14 Aug 1996 21:50:48 -0800
To: hfsig@tapr.org
Message-ID: <199608150501.WAA00823@PEAK.ORG>
Rendered body type: text/plain

Phil,

Thanks for the feedback. You make some good points here and your follow-up
is even more interesting and noteworthy. I'll comment on that shortly.

>>I wonder what a suitable ECC code would be for a short bit field such as
>>32-bits.
>
>Is that 32 data bits plus however much overhead is necessary?

Right, 32 bits data + x coding bits; whatever it takes.

>
>Why use such a small block? Performance is always better for larger
>blocks.

Granted; The header is as important as correct synchronization. Without
that, the message cannot be run through the Fano decoder as one would'nt
know how the bit stream is to be composed. Typically, the block will be a
very compact field.

>
>The more I think about the problems of a fading channel, the more I am
>troubled by schemes that separately encode sync, header and
>data. There's something to be said for a scheme that encodes and
>interleaves a continuous bit stream and then layers a simple packet
>framing scheme on top that doesn't have to deal with frequent errors
>(e.g., HDLC). Then the packet header would enjoy as much protection
>as a much larger data field.

This is a concern; With flat fading, or certain kinds of selective fading,
this certainly may be valid. The sync vector/header, as presented, does not
really exploit time diversity to address fading. Except (and that is part of
the question I posed), if the right kind of code is used, i.e., one that
effectively deals with burst errors, like for example RS.

The proposed scheme does however, offer some strenghts to this problem. It
does exploit a certain degree of frequecy diversity in addition to robust
modulation waveform for this portion; each of the four channels carries the
same information (replicated x 4) in four BPSK identical channels. These two
factors, together, offers some advantages that are put to good use to combat
the HF channel. If I may illustrate the expected gain in error performance
that this robust header may offer:

                For a BER of 10-5 in a 3 kHz BW channel:
1200 bps with 2 fading channels, 2ms multipath, and 1 Hz fading BW req. 11
dB S/N
  75 bps with 2 fading channels, 5ms multipath, and 5 Hz fading BW req. 2
dB S/N
                                                                      ^^^^^

(from MIL-STD-188 documents - some liberties taken)

We are looking at the 75 bps performance figure for the sync vector and
header portion. Think we can agree that this illustrates a fairly realistic,
though poor HF channel, and that the encoding scheme/waveform is not doing
too bad, i.e. 2dB S/N for a BER of 10-5.

--Johan

108. Re: [HFSIG:1504] Re: ECC for SHORT data block

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 15 Aug 1996 03:15:23 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608151015.DAA19279@servo.qualcomm.com>
Rendered body type: text/plain

Some further reflections on the scheme I sketched out earlier today.
My original reason for using a convolutional interleaver was to make it
practical to try all possible interleaver phases in the acquistion phase.

But that's not necessary if I mux in a continuous PN sync
sequence. Just make the PN period equal to the length of a block, and
phase it accordingly. Then when you find the peak from the PN
correlator at the receiver, you know exactly where all the symbols are
in the block and where the block begins and ends.

This gets rid of all overhead from filling and flushing the
interleaver pipeline, since a block interleaver produces exactly as
many symbols as it gets. But it does mean using a fixed size block,
and there may be some internal fragmentation (fancy term for "unused
space") on small packets. But again, this may be inevitable if you're
trying for enough time diversity to overcome fading.

Phil

109. TAPR offers Motorola EVM56002 group purchase

From: Greg Jones <wd5ivd@tapr.org>
Date: Fri, 16 Aug 1996 12:22:26 -0500 (CDT)
To: tapr-bb@tapr.org (TAPR-BB mailing),
        amsat-bb@amsat.org (AMSAT BB Mail Group),
        hfsig@tapr.org (HF SIG mailing), dsp-93@tapr.org (DSP-93 Build)
Message-ID: <199608161722.MAA09958@tapr.org>
Rendered body type: text/plain

TAPR offers Motorola EVM56002 group purchase   8/15/96

TAPR and Motorola have agreed on a special group purchase of the Motorola
DSP56002 Evaluation Modules (EVM) for a limited time. The DSP56002 EVM is a

general purpose DSP board with on board peripherals and software for assembling and debugging code. TAPR would like to thank Tim Baggett, AA5DF, of Motorola for facilitating this purchase.

The price will be:
   $85.00 US (+ s/h) **

         The EVM56002 normally retails through distributors for $150.

         ** No 10% Discount for TAPR Members.
            Due to the nature of the purchase, TAPR cannot offer a
            membership discount.

For more information on the DSP56002EVM purchase, please see the TAPR web page at http://www.tapr.org

For more information on the DSP56002 EVM and Motorola DSP products visit the Motorola DSP web page at http://www.motorola-dsp.com.

Please note: This unit provides a minimum interface for radio connections. Additional circuits are required for specific radio interfaces based on the application. See the TAPR HF SIG Upload Area (ftp.tapr.org /tapr/SIG/hfsig/upload) for files on one type of radio interface. There are other units that provide complete radio interfaces and should be considered if you feel that you are not capable of building such items.

This EVM group purchase will include:

     * EVM56002 Board
        See below for specs.

     * Motorola's DSP56000 cross assembler
     * Domain Technologies' debug GUI debug software (DOS, >386 CPU)
        both communicates with the EVM over an RS232 serial port

     * Available amateur software developed for the 56002 EVM board

     * Documentation
          56000 Family Manual,
          56002 User's Manual,
          56002 Data Sheet,
          Quickstart Guide,
          Debugger manual, and
          Schematics

TAPR will be taking orders for up to 200 units.

Orders will be taken through the ARRL/TAPR Digital Communications Conference in Seattle on September 21-22, 1996.

Shipping will not begin until the units are delivered from Motorola, which will be sometime after September 22nd. Money will be deposited when the order is placed with Motorola.

If you would like to order one of these units, you can
      e-mail tapr@tapr.org,
      phone (817) 383-0000, or
      fax (817) 566-2544.

      Office Hours: Tuesday - Friday 9am - 12pm, 3pm - 5pm Central Time

Questions concerning the unit and details on the buy will be handled on the TAPR DSP-93 Special Interest Group list. To subscribe, send e-mail to listserv@tapr.org. In the message type subscribe dsp93 YourFirstName YourLastName. The server will then send you a message back. Announcements on the status of the shipment will be made to the TAPR DSP-93 list and TAPR-BB lists.

Shipping and Handling
----
* Shipping and Handling within the US will be $10.00 US

by UPS Ground unless otherwise requested by purchaser.

* International Shipping will need to contact the TAPR office and get a quote on the shipping to your country. TAPR uses International Express Mail, unless the purchaser requires something else.

General Information

The DSP56002EVM is based on the 56002 general purpose DSP by Motorola. The 56002 is a 24-bit precision, dual memory space harvard architecture digital signal processor. The board contains a stereo CD-quality audio codec, on-board 32k words external SRAM, and an expansion spot for a 32K flash EEPROM. The user only needs to supply the IBM PC compatible computer, a 7-9 volt AC or DC power supply, and an RS232 cable for interfacing to the PC.

The DSP56002 EVM is perfect for the hobbyist and student wishing to learn DSP programming on the Motorola 56002 DSP. Many amateur radio applications have been written and used on the DSP56002 EVM including, 1200 AFSK packet, 9600 FSK packet, RTTY, BPSK, QPSK, and various audio filters for CW, SSB, and QRM/QRN reduction.

The 24-bit DSP56002 feature:
- 40 MHz operation (new EVM's are 6 MHZ, and most have operated at much higher clock rates)
- 20 MIPS at 40 MHz (50 nS instruction cycle at 40 MHz)
- 120 Million Operations per second (MOPS) at 40 MHz
- 512x24 bit on-chip program RAM with bootstrap ROM
- two 256x24 bit on-chip data RAMs (X and Y data spaces)
- External memory expansion with 16 bit address and 24 bit data bus
- Byte wide host interface

- Synchronous Serial Interface (SSI)
- Serial Communications Interface (SCI)
- 24-bit timer/event counter
- software programmable internal clock PLL

The EVM features:
- 32kx24 bit zero wait-state external SRAM
- option for 32Kx8 flash EEPROM for program bootstrapping and stand-alone operation
- Crystal Semiconductor's CS4215 stereo CD quality sigma-delta A/D D/A
- on-board voltage regulator
- RS232 interface to the OnCE (tm) controller
- RS232 interface to the DSP SCI port for communication between the DSP56002 and a terminal (PC) directly
- connectors for external access to the DSP56002's memory expansion, host interface, and serial interfaces (SSI, SCI)
- documentation (56000 Family Manual, 56002 User's Manual, 56002 data sheet, quickstart guide, debugger manual, schematics)

The on-board Crystal A/D D/A features:
- 24.576 MHz crystal for sampling at 48, 32, 16, 9.6, and 8 Ksps
- 16 bit linear, 8-bit mu-law, 8-bit a-law, and 8-bit linear sampling

- the 24.576 MHz crystal can be replaced to provide other sample rates
- Software programmable signal attenuation

P.S. TAPR is also doing a group purchase on PC-DSP and PC-SIM for Windows. This is an excellent package for learning and developing DSP software. See: http://www.tapr.org

The PC-DSP is an interactive, menu-driven software package used for: waveform synthesis using a variety of methods, basic signal operation, fast Fourier transforms, convolution and correlation, solution of difference equations, analysis and design of IIR and FIR filters, digital filter simulation and code generation, and power spectrum estimation using classical and modern techniques. Some key features of PC-DSP listed include: GNUPLOT support, code generation, macro compiler, dialog compiler, sound file support, data file formats, and compatibility with PC-SIM.

Tucson Amateur Packet Radio
8987-309 E Tanque Verde Rd #337 * Tucson, Az * 85749-9399 * 817-383-0000
---
e-mail: TAPR@TAPR.ORG ftp: ftp.tapr.org web: http://www.tapr.org/tapr
---

110. [HFSIG:1500] Re: Sound Cards -Reply

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Tue, 20 Aug 1996 07:42:39 +0200
To: hfsig@tapr.org
Message-ID: <s2196c2b.055@KIDD.CO.ZA>
Rendered body type: text/plain

Johan commented:
The other alternative is to look for a Turtle Beach Tahiti card - that one though has a 56002. There is a software toolkit available for that one as well. It's a really nice PC plugin card.

Hope this helps.

--Johan

How much is this card more or less Johan ? Does it have a debugger ?
danie

111. Re: [HFSIG:1507] Re: Sound Cards -Reply

From: forrerj@peak.org (Johan Forrer.)
Date: Tue, 20 Aug 1996 08:47:32 -0800
To: hfsig@tapr.org
Message-ID: <199608201556.IAA16578@PEAK.ORG>
Rendered body type: text/plain

Danie,

>Johan commented:
>The other alternative is to look for a Turtle Beach Tahiti card - that one
>though has a 56002. There is a software toolkit available for that one as
>well. It's a really nice PC plugin card.
>
>Hope this helps.
>
>--Johan
>
>How much is this card more or less Johan ? Does it have a debugger ?
>danie
>
>

Don't know the price, but to be competitive, it would be in the $250 range. And yes; there is a 3-rd party software toolkit (not from Turtle Beach) with a debugger that apparently has the usual breakpoint/single step debugging.

While on the subject: if anyone would like to tackle the DSP chip on the Creative Labs. "Soundblaster" boards, i.e., the chip they call the "ASP", I have found the manual for it on the Web. Unfortunately it is a rather limited DSP - better to be left alone (my opinion). Anyway, CL have lately been using 56002's in their video conferencing products.

There also is a TI320C32-based sound card that may interest you; see www.core-dynamics.com for further details. Not sure what the deal is regarding your own software development for it.

A bit of feedback on the TI320C31 floating point DSP that may interest you: That turned out to be a somewhat of a dissapointment for me; the multiplier takes two 32-bit integers and the result is a 40-bit number. Compare this with the 56002 EVM that takes two 24-bit (or 48-bit doubles) but gives you 56-bit accumulator arithmetic results. Or the ADSP 21xx family that takes two 16-bit data words and results are 40-bit numbers. It is possible of course to achieve extended precision through multiple instructions, albeit at a performance hit. Unless one really needs 32-bit floating point multiplies, I think you will agree that the 56002 EVM is an exceptional bit of hardware.

Sorry - nothing to do with HF digital, but hope some useful information for experimentors.

--Johan

112. Re: [HFSIG:1495] FTP Packet Driver Specification

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 20 Aug 1996 19:37:04 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199608210237.TAA17725@servo.qualcomm.com>
Rendered body type: text/plain

Johan,

For the FTP Packet Driver spec, see

http://www.crynwr.com/crynwr/packet_driver.html

Note that all this really does is to provide a portable interface to
an Ethernet card. Although the spec is supposedly general enough to
support other kinds of interfaces, I don't know that it has ever been
used for this purpose.

The software that calls a packet driver must still know quite a bit
about the Ethernet link layer, such as addresses, type fields, and the
like. So unless you are prepared to make you modem look just like an
Ethernet board (which is a questionable thing to do for an HF radio
modem) you don't really want to use this interface. What you really
want is a full-blown IP router that can encapsulate and decapsulate IP
in various link layer protocols such as AX25, Ethernet and whatever
link protocol you create for your modem. That would be much more
useful, I think.

Phil

113. Re: [HFSIG:1508] Re: Sound Cards -Reply

From: rickc@tridelta.com (Rick Contaldo)
Date: Wed, 21 Aug 1996 09:01:18 -0400 (EDT)
To: hfsig@tapr.org, hfsig@tapr.org
Message-ID: <199608211301.JAA01480@sparcy.tridelta.com>
Rendered body type: text/plain

At 11:01 AM 8/20/96 -0500, Johan Forrer. wrote:

>While on the subject: if anyone would like to tackle the DSP chip on the
>Creative Labs. "Soundblaster" boards, i.e., the chip they call the "ASP", I
>have found the manual for it on the Web. Unfortunately it is a rather
>limited DSP - better to be left alone (my opinion). Anyway, CL have lately
>been using 56002's in their video conferencing products.
>
>
>--Johan
>
>

Johan,

I'm curios about the Creative Labs ASP. Would you please point me to
the site where you found the manual?

Thanks,

Rick

114. Frame-Check Sequence for AX.25

From: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Date: Wed, 21 Aug 1996 14:12:34 -0400
To: hfsig@tapr.org
Message-ID: <199608211812.OAA04139@appstate-01.campus.mci.net>
Rendered body type: text/plain

I'm trying to code sending packet info and for the Frame Check Sequence
(FCS) I find the AX.25 specs only say, "It shall be calculated in accordance
with ISO 3309 (HDLC) Recommendatons." Anyone know where I could find a
description for the FCS calculation (or even better Motorola 68HC11 code!).

Don, K4JPJ

115. Re: [HFSIG:1510] Re: Sound Cards -Reply

From: Johan Forre <forrerj@peak.org>
Date: Wed, 21 Aug 1996 12:28:00 -0700 (PDT)
To: Rick Contaldo <rickc@tridelta.com>
Cc: hfsig@tapr.org

Message-ID: <Pine.SUN.3.91.960821122524.23532A-100000@kira>
Rendered body type: text/plain

Rick,

On Wed, 21 Aug 1996, Rick Contaldo wrote:

> At 11:01 AM 8/20/96 -0500, Johan Forrer. wrote:
>
> >While on the subject: if anyone would like to tackle the DSP chip on the
> >Creative Labs. "Soundblaster" boards, i.e., the chip they call the "ASP", I
> >have found the manual for it on the Web. Unfortunately it is a rather
> >limited DSP - better to be left alone (my opinion). Anyway, CL have lately
> >been using 56002's in their video conferencing products.
> >
> >
> >--Johan
> >
> >
>
> Johan,
>
> I'm curios about the Creative Labs ASP. Would you please point me to
> the site where you found the manual?

Look for the D-950 core under chips at SGS-Thompson's web site (www.st.com).

>
> Thanks,
>
> Rick
>
>
>

Good luck.

--Johan

116. Re: [HFSIG:1509] Re: FTP Packet Driver Specification

From: forrerj@peak.org (Johan Forrer.)
Date: Wed, 21 Aug 1996 19:38:49 -0800
To: hfsig@tapr.org
Message-ID: <199608220248.TAA04371@PEAK.ORG>
Rendered body type: text/plain

Phil,

Thanks for the pointer and also thanks to all the others that pointed me in
the right direction.

With Pawel's help, I hacked a version of his AX25 low level driver and then
have NOS call the driver through a software interrupt via standard FTP
packet driver call mechanism. Looked simple enough (Heck, I learned a lot.)

I have been testing and playing with this software interface for monitoring
300 baud HF packet, although I have no intentions to use 300 baud HF packet
- just thought it a good test. NOS is used to monitor AX25 traffic, but I
could have used a number of user interface programs that would accept the
interrupt hook.
It is encouraging to see how well the low-level HDLC code seems to do,
considering the poor conditions and very low signal levels on the bands.
However, I must admit that the DSP modem is probably the main contributing
factor.

Anyway, still a lot more to do and a lot more to learn.

73's

--Johan

117. PSK Sound files

From: forrerj@peak.org (Johan Forrer.)
Date: Fri, 22 Aug 1997 10:46:48 -0800
To: HFSIG@tapr.org
Message-ID: <199608221755.KAA20435@PEAK.ORG>
Rendered body type: text/plain

Hi folks,

I uploaded some .WAV sound files from my experimental DQPSK
modems. These will give you an idea what the waveforms sound
and look like. Included are single and multi-channel m-DPSK
samples. The package is available for downloading from:

ftp://ftp.tapr.org/SIG/hfsig/upload/psk_snd1.txt and psk_snd1.zip

Wish I had more time to work on this project, but things have
been busy as usual.

73's

--Johan

118. PSK Sound files

From: forrerj@peak.org (Johan Forrer.)
Date: Fri, 22 Aug 1997 23:49:43 -0800
To: hfsig@tapr.org
Message-ID: <199608230658.XAA05524@PEAK.ORG>
Rendered body type: text/plain

Hi,

If you are having problems locating the PSK sound files, either
try:

http://www.tapr.org using a web browser - follow the links to hfsig/upload, or;

FTP directly to: ftp.tapr.org/tapr/SIG/hfsig/psk_snd1.txt + .zip

That should do it. Sorry about that.

--Johan

119. New member

From: vk7zo@calvados.apana.org.au (Graham Ranft)
Date: Fri, 23 Aug 96 09:07:22 +1000
To: hfsig@tapr.org
Message-ID: <Za39uD2w165w@calvados.apana.org.au>
Rendered body type: text/plain

Hi just a short one to say hello and introduce myself on HFSIG.
Recently aquired a paccom ptc level 1 and this has started me looking
again at digital sig precessing and digging into coding etc as well as
things ionospherical... I am currently only operation on 3632 owing to
noise problems on 20 mx from ptc which is going to require a complete
station rework-earthing etc etc probably plus ferrite beading and other
shutting up measures. Since I am going to be moving qth before to much
longer this can wait till then. I have been active on pacsats and 9600 bd
sats this is a new and welcome (and interesting) change in direction.
If anybody has any mods to improve paccom ptc 1's I would be grateful for
any info.
I have interent WWW access and have accessed Phil Suusmans page (and
subscribed) as well as SCS .
Hope to meet some of you via this or on HF 'ere long-be hapy and well
Graham at Dromedary near Hobart Tasmania , Australia.
email vk7zo@ calvados.apana.org.au
packet vk7zo@vk7gl.#hbt.tas.oc.au
73's

--
vk7zo@calvados.apana.org.au (Graham Ranft)
APANA, Hobart, Tasmania: 03 6234 8530

120. Re: [HFSIG:1457] Re: EVM56002 ?

From: Greg Jones <wd5ivd@tapr.org>
Date: Sun, 25 Aug 1996 10:28:02 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199608251528.KAA02689@tapr.org>
Rendered body type: text/plain

Yes -- this is a great offer. Dorothy has about 40 on order right now.

Please pass the word to students and school, since they can get in on the
offer as well.

I am hoping this is one of many future involvements with Motorola in the
future.

Cheers - Greg, WD5IVD

>
> Greg Jones wrote:
>
> > There is an opportunity for TAPR to do a group purchase on the Motorola
> > EVM56002 at a pretty low cost. Between 20 and 200 units would be the
> > goal...same price no matter how many are purchased. Figure about $60 under
> > current commercial prices. No hard prices, becuase more work would have to
> > be done.
> >
> > Would there be interest in this in the HF SIG ? Espiecally since Johan's
> > code would run on it.
>
> Sounds great considering the price is already only $149.95 at the
> Design-Net price page (http://www2.motorola-dsp.com/dsp/home/net/pg.html).
>
> They also offer a student discount of 20% reducing the price to $120 + $3
> shipping (http://www2.motorola-dsp.com/dsp/home/edu/discount.html)
>
> Mike N4CNW
>
>

121. Re: [HFSIG:1517] Re: EVM56002 ?

From: forrerj@peak.org (Johan Forrer.)
Date: Sun, 25 Aug 1996 21:30:58 -0800
To: hfsig@tapr.org
Message-ID: <199608260439.VAA01385@PEAK.ORG>
Rendered body type: text/plain

Hi Greg,

Sounds like things are going with the EVM orders - I'm pleased to hear that.
I don't think anyone will be dissapointed as it is a great DSP to work with;
a lot of PD code and development tools available too. I suspect that by time
of the DCC you may even have further orders.

Looking forward to the DCC. My wife will be coming along this time, and we
will be in Seattle already by Thursday to visit a bit. One of our favorite
cities.

I need to mention one item for the HFSIG agenda that we have to take care of
at the DCC; that is that we need to find someone that would be willing to
fill the function of moderator/scribe/organiser. Perhaps it is a good idea
to get some fresh ideas. I will continue my participation and sharing in my
experimental and development efforts.

--Johan

>Yes -- this is a great offer. Dorothy has about 40 on order right now.
>

>Please pass the word to students and school, since they can get in on the
>offer as well.
>
>I am hoping this is one of many future involvements with Motorola in the
>future.
>
>Cheers - Greg, WD5IVD

122. Who sells HF path simulators and related equipment?

From: Rob Roy <rroyael@laser.net>
Date: Tue, 27 Aug 1996 13:26:35 -0500
To: hfsig@tapr.org
Message-ID: <2.2.16.19960827182635.315fb6f4@smtp1.laser.net>
Rendered body type: text/plain

Hi;

  I realize that there are a lot of different HF path simulators
that are used for testing. I would like to obtain any and all
information on current/future systems. The results of this
query will be posted.

--
                    Rob Roy
rroyael@laser.net

123. Re: [HFSIG:1519] Who sells HF path simulators and related equipment?

From: forrerj@peak.org (Johan Forrer.)
Date: Tue, 27 Aug 1996 13:52:09 -0800
To: hfsig@tapr.org
Message-ID: <199608272100.OAA01809@PEAK.ORG>
Rendered body type: text/plain

Rob,

I cannot speak for professional equipment manufacturers - there are several
and they are expensive ~ $30,000 range for a Watterson channel simulator.

However, a blatant plug for the work and interests from this list (HFSIG) -
we are proud of it.

HFSIG have been discussing HF channel simulators based on the Watterson
ionospheric model for some time (approximately 3 years). This model
represents narrow-band ionospheric propagation behavior as it affects HF,
typically an SSB bandwidth channel (ca. 3000 Hz) - typically to conform to
test conditions as set out by the CCIR. This includes various types/degrees
of channel disturbances, such as multipath, flat, and selective fading.
Although of much importance, this does not include simulated noise - that is
the subject of further work.

The objective and incentive for this work was to provide a low-cost
implementation for amateurs to evaluate modems for HF use. Based on work
initiated by Alexander Kurpiers, DL8AAU, and with generous assistance from
Juergen Hash, DG1SCR, I made a simplified version of such a simulator. This
particular implementation runs on TAPR's DSP-93, which uses a TMS320C25 DSP
chip. Executable code is available for downloading (free for amateur use) on
the HFSIG files area. Contact TAPR for further details on the DSP-93
hardware. It may be a good idea to get in touch with Alexander should you
want to know more about this particular implementation.

You may also want to browse through the work done by Eric Johnson for NTIA
regarding ALE (atanasoff.nmsu.edu/pub/hf). I have been told that Eric does
related consulting work and offers a low-cost HF channel simulator.

As for the future - I am considering developing a professional version HF
channel simulator to compete with the $30,000 Watterson channel simulator
hardware. E-mail me for further information if you are interested.

Hope this is of interest.

--Johan Forrer
(e-mail: forrerj@peak.org)

>Hi;
>
> I realize that there are a lot of different HF path simulators
>that are used for testing. I would like to obtain any and all
>information on current/future systems. The results of this
>query will be posted.
>
>--
>   Rob Roy
>rroyael@laser.net
>
>
>

124. HF Channel Simulator

From: ejohnson@NMSU.Edu ()
Date: Wed, 28 Aug 1996 09:01:10 -0600
To: hfsig@tapr.org
Message-ID: <199608281501.JAA18442@NMSU.Edu>
Rendered body type: text/plain

Hi,

Tim Baggett (a former student) forwarded your request for info about
HF channel simulators. By mid-September I'll be shipping a new
low-cost Watterson model simulator based on the Motorola 56002.

It simulates two channels simultaneously, each having two independently-fading paths. The user interface is a serial port, through which you can set (for each channel):

* the SNRs for the two paths
* the Doppler spread (fading bandwidth)
* the delay between the two paths.

Since this was originally developed for manufacturers of MIL/FED-STD ALE gear, there is also a set of preset combinations for Gaussian noise, ITU-Good, and ITU-Poor channels at specific SNRs.

The selling price for just the DSP board (unencumbered by case, power supply, etc.) is $500. A unit mounted in a case with power supply is $1000.

Contact: Johnson Research
100 Horseshoe Circle
Las Cruces, NM 88005

Regards

Eric Johnson